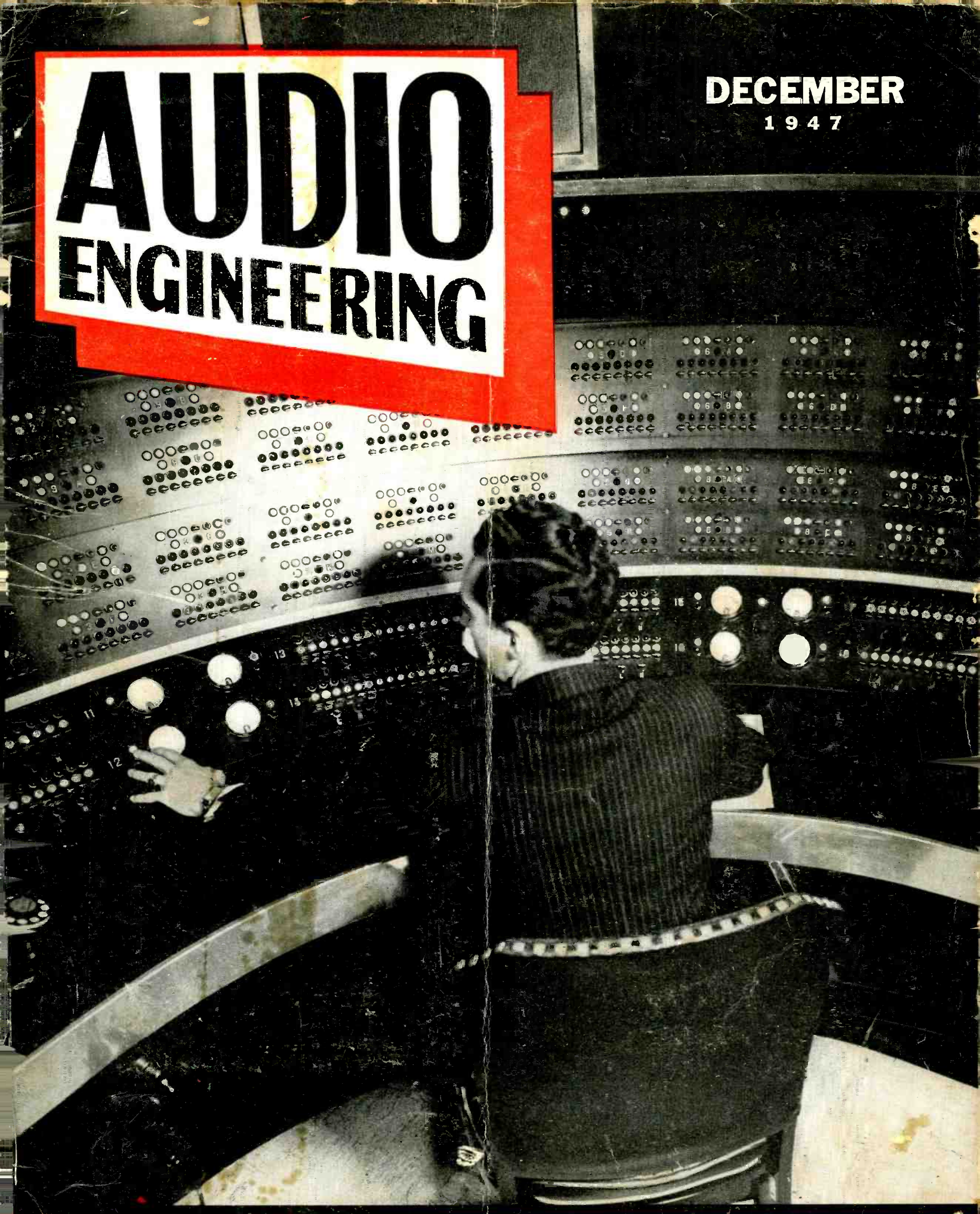
AUDIO ENGINEERING

DECEMBER
1947



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Vol. 31, No. 11

COVER ILLUSTRATION

Master Control Board, WNBC.

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EDITOR'S REPORT

MORE ABOUT NOISE SUPPRESSORS

● JOHN D. GOODELL, whose excellent article on the Scott Noise Suppressor appeared in the November issue of this magazine, has written in asking us to inform our readers that the model 910A dynamic noise suppressor for broadcast station use is manufactured only by Hermon Hosmer Scott. Since the article did not so state, he has asked us to give this point special emphasis. A letter from Goodell on this subject is published in our "Letters" column. The simpler six-tube version, also described in Goodell's article, is manufactured by Goodell under license from Scott and forms a part of amplifiers and other radio apparatus produced by the Goodell organization.

Goodell adds that he is now using a 6B8 input tube in the six-tube version, replacing the 6SQ7 shown in his original diagram. A revised diagram will appear next month.

Complete details on the experimental noise suppressor developed by the General Engineering department of the American Broadcasting Company have just been received, so we are publishing the schematic diagram showing all parts values in this issue. See page 28. This is the noise suppressor described by John D. Colvin of ABC at the recent convention of the National Association of Broadcasters in Atlantic City. It is based on fundamental circuits developed by Dr. Harry D. Olson of RCA, which have already been published elsewhere, but this is the first practical working model with parts values to appear in any magazine. A complete article on this noise suppressor, by Charles D. Cole of the American Broadcasting Company, is scheduled for our January issue.

THE MUCH-MALIGNED AUDIO TRANSFORMER

● ONE of the most prevalent erroneous ideas is that resistance-coupled audio amplifiers make possible far better reproduction than can be obtained by using audio transformers for the purpose. In fact, some engineers who really ought to know better dream of output tubes of such low output impedance that they may be coupled directly to loudspeaker voice coils, thus eliminating the output transformer. But if we keep in mind that a radio receiver is but one link in a circuit which starts at the broadcasting microphone, the possible degree of improvement which may be effected in this manner becomes very small indeed, especially if the program is being picked up from some remote point. Most broadcast-type microphones employ coupling transformers to the preamplifiers which, in turn, are coupled by line transformers to voltage and power amplifiers, thence, again by transformers, to the transmitter output tubes. In the course of its peregrinations, it is not unusual for a broadcast signal to wander through possibly two hundred audio transformers before it eventually arrives, in demodulated form, at the input to the audio amplifier of a radio receiver. Thus, whether we use or do not use two or three more audio transformers in the amplifier it can make no appreciable difference in the ultimate reproduction.

Of course, the audio transformers used must be good—at least as good as those used in the broadcast system amplifiers—otherwise the fidelity of reproduction must suffer. But with good transformers, there can be no perceptible advantage insofar as fidelity of reproduction is concerned by substituting some other means of coupling. Resistance coupling is cheaper, of course, but not so reliable; otherwise the telephone companies would most certainly not invest in expensive transformers. And for those applications of audio amplifiers in which the audio signal is fed directly from a phonograph pickup, microphone, or other form of transducer, we doubt if anyone can hear the slight inevitable distortion present in good transformers. If we could, then the cumulative effect of the number used in broadcasting would make even the best radio program unbearable.

WITH OUR AUTHORS

● OUR Record Revue department has been taken over this month by Bertram Stanleigh, who will be a regular contributor hereafter, sharing space with Edward Tatnall Canby. Mr. Canby couldn't make the deadline this month, but his excellent column will continue regularly in the future . . . Winston Wells is recovering from a long siege of illness and will shortly resume his series on the design of electronic organs . . . S. Young White will be back next month with another of his outstanding articles on ultrasoncis . . . George Nixon of the National Broadcasting Company has just completed an excellent article which is now being cleared for publication . . . Dr. Wetzel has finished his last instalment of his series on magnetic tape recording theory, which will appear next month . . . and a Merry Christmas and Happy New Year from all of us to all of you.—*J. H. P.*

Letters

Audio Association?

Sir:

After receiving the first few issues of your new magazine, I must say that I believe you are serving a much neglected field, in a highly adequate manner.

Now that the audio engineer has been dignified by a specialized publication, which will tend to draw the members of the field together, is it not time for him to have an organization of his own?

I have in mind an association similar in function and purpose to the I. R. E. and S. M. P. E. in their respective fields.

I will be glad to correspond with anyone interested in this matter.

Frank E. Sherry, Jr.
705½ W. San Antonio St.
Victoria, Texas

What do our readers think?—Ed.

Scott Noise Suppressor

Sir:

In the Dynamic Noise Suppressor article which appeared in your November issue, the diagram of the 910A broadcast version of these units was included, but I failed to designate the manufacturer.

Our organization manufactures the Dynamic Noise Suppressor incorporated in the Goodell Radio Phonograph and the Goodell Dynamic Noise Suppressor Amplifier. The latter is licensed only for use in phonograph and phonograph distribution systems. The broadcast models designed especially for broadcast station application are manufactured exclusively by Hermon Hosmer Scott, Inc., in Cambridge, Mass.

John D. Goodell, President
The Minnesota Electronics Corporation
St. Paul, Minn.

Correction

Sir:

In the Technicana Section of the October, 1947 issue, there are errors in the third paragraph of your summary of my paper, "Decay Rates of Piano Tones."

1) The rate of decay of conventional piano tone is not, as you have implied, dependent upon the striking force.
2) The decay rates of the electronic piano tones were shown to be relatively independent of the voltage gain of the amplifying system, rather than the amplitude.

D. W. Martin
R. C. A. Victor
Advanced Acoustic Dev'l.
Camden, New Jersey

Acoustical Array

Sir:

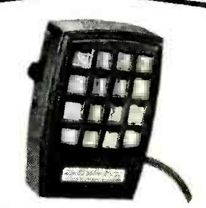
While on an assignment in a Naval shipyard during the war, we noted a rather weird "dead" area near a stores

[*Continued on page 39*]


ON E-V DESK STAND
Mounts on new E-V Model 415 reclining desk stand


ON FLOOR STAND
Has 5/8"-27 thread adapter for conventional stands


HAND-HELD
Fits hand comfortably


AVAILABLE WITH SWITCH
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Size is 3"x 2$\frac{3}{16}$"x 1". Crystal model weighs only 6 ounces. Highest purity diecast metal case, finished in lustrous gray-brown.

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Model	Type	Output db	Freq. Resp. cps	Cable	Price
915	Crystal	—50	60-7500	7 ½ ft.	$10.00
915-S*	Crystal	—50	60-7500	7 ½ ft.	11.50
615	Dynamic	—57	55-7500	7 ½ ft.	16.50
615-S*	Dynamic	—57	55-7500	7 ½ ft.	18.00
615-SR**	Dynamic	—57	55-7500	7 ½ ft.	20.00
215	Carbon	—26†	200-4000	48 in.	8.25
215-S*	Carbon	—26†	200-4000	48 in.	9.75
215-SR**	Carbon	—26†	200-4000	48 in.	10.00

*with slide-to-talk shorting switch
**with slide-to-talk relay control switch and hang-up hook
†using input transformer

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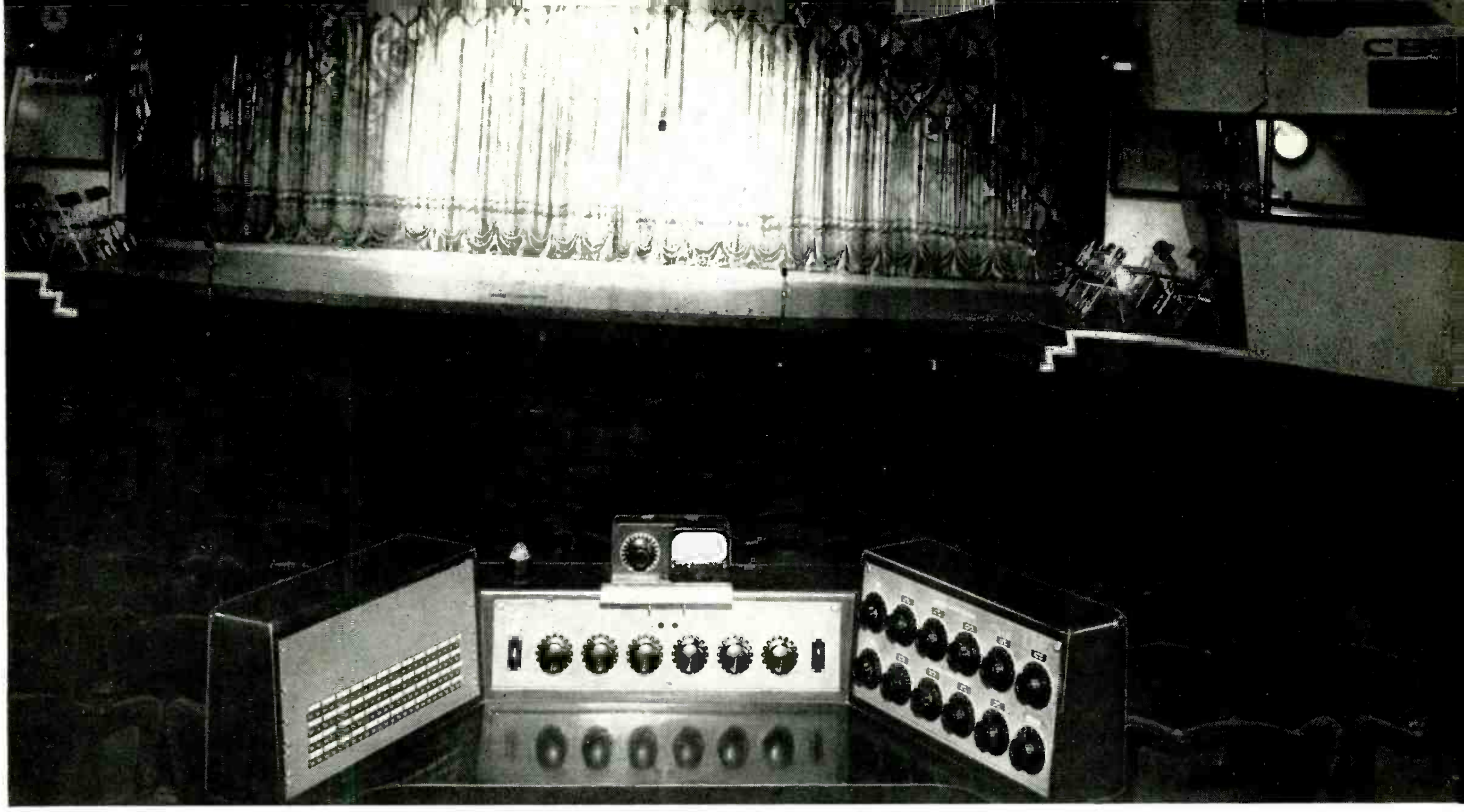


Fig. 1. A sound reinforcement control console installed in the audience area of a CBS radio theater. Such a vantage point is highly desirable for this activity as the operator has first-hand knowledge of the degree of reinforcement required and the effectiveness of all operating adjustments.

Broadcasting Studio Sound Reinforcement

HOWARD A. CHINN* and ROBERT B. MONROE**

How special public address problems encountered in a broadcasting studio may be solved.

IN LARGE broadcasting studios and radio theaters, where a studio audience is present during the presentation of radio programs, it is usually essential to provide sound reinforcement (PA) facilities to enable the studio audience to hear all portions of the program in proper perspective. If these facilities are not provided, the undesirable situation often arises where the studio audience is unable to hear parts of the program. In some types of programs, such as an audience participation show, this may result in an otherwise outstanding performance falling "flat."

In addition to reinforcing sounds originating on the stage of the broadcast studio, the sound reinforcement system also serves other purposes. For example, by means of the sound reinforcing equipment, the studio audience is enabled to hear those portions of a program which may originate from a remote point. Another use of the sound reinforcement equipment is in achieving special effects, such as the illusion of added room reverberation, or the creation of echo effects.

*Chief Audio Engineer, Columbia Broadcasting System, Inc.

**General Engineering Dept., Columbia Broadcasting System, Inc.

Each of these functions of the equipment must be kept in mind in designing a studio sound reinforcement system.

Design Consideration

A simple (but ineffective) method of obtaining the program feed for sound reinforcement is to bridge the sound reinforcement amplifier, or amplifiers, across the outgoing studio program line. When this is done, the program material reproduced on the sound system is exactly the same as that leaving on the outgoing line. Such an arrangement, while workable, leaves much to be desired as all portions of the program are then reinforced equally. More effective sound reinforcement is obtained, on the other hand, when only those portions of the program which could not otherwise be easily heard are reproduced on the studio loudspeakers. As an example, an orchestra which can be heard without difficulty in all parts of the studio need not be reinforced and to do so would result in a jumbled, confusing performance reaching the studio listeners. However, the voice of a featured singer performing with the orchestra should be reinforced to be easily heard in all parts of the studio.

From the above discussion it can be seen that satisfactory sound reinforcement requires a different mixing technique from that employed in mixing program material for the usual broadcast purpose. This is often accomplished by employing a separate mixing console and operator for the specific purpose of preparing the program material for sound reinforcement and properly controlling the over-all loudspeaker levels during the performance.

Methods Employed

It is obvious that it is not practical, nor desirable, to employ separate microphones in the studio for sound reinforcement purposes. It is therefore necessary to design the audio system in such a manner that the output voltage from each of the studio microphones is available at both the regular mixer and sound reinforcement mixer. A method of accomplishing this, which has been in use by CBS for some time, makes use of a bridging transformer connected across the output of each of the microphone preliminary amplifiers (i. e., ahead of the regular mixer controls) as shown in *Fig. 2* The outputs of these transformers provide the program feeds to the sound reinforcement mixer.

In addition to supplying program feeds to the sound reinforcement mixer from each microphone channel, similar feeds should be provided from all other sources of program material. For example, if program material is obtained from transcription turntables or from incoming program loops, feeds to the sound reinforcement mixer should be arranged from these sources of program material. It is by means of the program feeds from incoming program loops that reproduction of program material from remote points is made possible.

Bridging pads may be used instead of bridging transformers, and accomplish comparable results at lower cost; however, due to the greater attenuation in the bridging pad, care must be taken that the signal level through the sound mixing system does not fall below the original microphone level if the greatest possible signal-to-noise ratio is desired. Another precaution to be observed when bridging pads are employed is the circuit grounding arrangement. If circuit grounds are made at more than one point in the system, difficulty with ground currents and resultant hum and crosstalk may be encountered.

It would not be economically justifiable to include a sound reinforcement mixer position for each of the program feeds, inasmuch as upwards of ten such feeds may be involved. An inexpensive and entirely satisfactory method of simplifying these mixing facilities is through the use of a smaller mixer panel, four positions usually being adequate for average installations; each individual mixer control receiving program material from a suitable selector such as a rotary switch or mechanically interlocked push button with the required number of positions, as illustrated in *Figs. 3* and *4*.

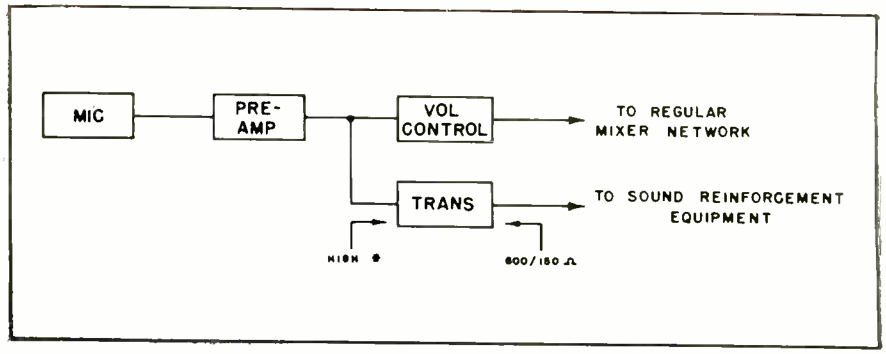


Fig. 2. Program feeds from the studio console to an external sound reinforcement mixer may be obtained by means of a bridging tansformer, or bridging pad, connected across the output of the microphone preliminary amplifier as shown. If bridging pads are used several precautions must be observed as discussed in the text.

In this way any of the program feeds may be connected to any desired mixer position.

The most suitable location for the sound reinforcement mixing console must also be given careful consideration. One possible location is in the control room immediately adjacent to the regular mixing console. This arrangement has been used in the past and has the advantage that all equipment is centralized in the control room and long low-level audio cable runs are not required. Such an arrangement has one very serious disadvantage; that is, the operator must depend on the studio loudspeaker level adjustments made before the program commences and has no way of knowing (with the possible exception of hand signals from an assistant in the audience) if the studio levels are satisfactory after the program gets under way. In view of this, it is generally desirable, despite the additional cable runs involved, that the sound reinforcement mixing console be installed in the audience section of the studio (see *Fig. 1*) where the operator will have first-hand knowledge of the degree of reinforcement required and the immediate result of all operating adjustments. When this is done, suitable communication (telephone or signal lights) should be provided between the sound reinforcement mixer and the control room to permit the control room to notify the sound reinforcement operator if the loudspeaker levels are interfering with the air program.

There is another method of controlling sound reinforcement which, although exhibiting one of the shortcomings already noted, has the advantage of extreme simplicity. In this method an additional operator is not required as the few controls involved may be mounted on the regular mixer panel and handled by the control room operator.

In this method, sound reinforcement feeds are obtained from each channel by means of a suitable bridging device. In this case, however, the bridging device is connected across the *output* of the regular mixer controls. The sound reinforcement feed thereby obtained then passes through a suitable "on-off" switch after which all feeds are combined by means of a differential network. The output of the differential network connects to the sound reinforcement amplifiers in the usual manner.

Operation of such a system is extremely simple. The operator simply selects the channels to be reinforced by means of the individual channel "on-off" switches. The level of each channel is, of course, a function of the adjustments being made on the regular mixer control inasmuch as the sound reinforcement feed is obtained at the *output* of this control. The only other control involved is a sound reinforcement master volume control which may be set prior to going on the air.

The disadvantage of this method lies in the fact that the operator, being in the control room, has no way of knowing the effectiveness of the sound reinforcement coverage after the program gets under way. Furthermore, the balance between the program elements being fed to the sound reinforcement system is determined entirely by the requirements of the air

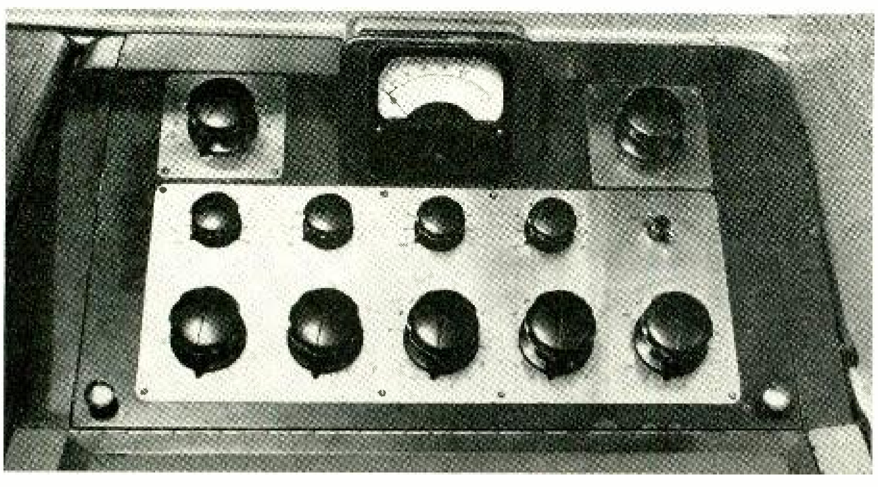


Fig. 3. A sound reinforcement mixing console of a type employed in several CBS studios. The circuit arrangement of this mixer is shown in Fig. 4. The four mixer controls are arranged across the lower row with the master gain control at the extreme right. Immediately above each mixer control are eight position rotary selector switches which permit program material from any one of eight sources to be connected to the mixer control. The twist key above the master gain control serves to connect a high-pass filter in the circuit. At the top of the panel a VI attenuator is located at the right of the volume indicator and a volume control for adjusting headphone monitoring level is at the left.

show. This will not necessarily be the optimum balance for the sound reinforcement service.

Loudspeaker Placement

The loudspeaker placement practices followed in motion picture theater sound systems and theater public address work cannot be employed in broadcast studio sound reinforcement because of the fact that microphones are often set up in the audience section of the studio. In some radio theater pickups, the main orchestra microphone is suspended as far back as the forward edge of the first balcony. For this reason, no general rules can be set down concerning the number of loudspeakers to be employed or their placement, as each particular installation presents specific problems.

In general, there are two different approaches to the problem of loudspeaker placement. The first makes use of a small number of loudspeaker units, usually two (but sometimes four) placed at each side of, and slightly forward of, the stage. This arrangement because of its simplicity and effectiveness, is in wide use at the present time. The second approach to the loudspeaker placement problem makes use of a multiplicity of loudspeaker units which are uniformly distributed throughout the audience area. Each loudspeaker unit is operated at a relatively low level with the result that the auditorium is literally flooded with sound although the level is not objectionably high at any one point. This arrangement can be used to advantage in theaters where a mezzanine and several balconies make coverage with a small number of speaker units impractical.

Acoustic Feedback

One of the more serious difficulties experienced in the operation of sound reinforcement systems is the tendency of the system to "sing" as a result of acoustic coupling between loudspeakers and microphones. This tendency to sing can be reduced by operating the loudspeakers at a reasonably low level as it is not necessary to provide the studio audience with more than comfortable listening level. In addition to conservative operating levels, there are several design considerations by means of which it is possible to reduce the tendency of a sound reinforcement system to sing. These design considerations include:

1) Loudspeaker units with sharp peaks in their response-frequency characteristics should be avoided as the high peak output at these points is often sufficient to set the system into oscillation even if it is operating at a conservative level. Loudspeaker units with smooth response-frequency characteristics, usually associated with the more costly units, reduce the tendency of a sound system to sing.

2) Inasmuch as the greatest energy content of average voice and speech exists in the lower audio frequencies (below 400 cps), the attenuation of these frequencies does much to reduce the tendency of the system to sing; furthermore, these frequencies may be attenuated without seriously reducing the intelligibility of reproduction. One method of reducing the low-frequency output consists of inserting a high-pass filter section in the system. A simple prototype filter section with a cut-off frequency in the range 250 to 400 cps and an attenuation in the order of 12 db per octave or less is usually effective. This filter should be removed from the system when reproducing program material from a remote point as wide-band reproduction is then desirable.

The above discussion suggests a simple, but very effective method of meeting the requirements of both (1) and (2) in a direct manner. This is through the use of dual loudspeaker units of the theater type which employ a multicellular high-frequency reproducer and a horn or direct radiator type low-frequency reproducer with a cross-over frequency of the order of 400 cps. When employing the system for sound reinforcement purposes, the high-frequency unit only is used. This type of reproducer usually has reasonably smooth response-frequency characteristics above the cross-over frequency, thereby meeting the requirements of (1) and has the required attenuation below the cross-over frequency, thus meeting the requirements of (2). When the system is employed for sound reproduction of program material from a remote point and full frequency range reproduction is desired, it is simply necessary to use the complete dual reproducer system.

The use of the high-frequency reproducers alone reduces the susceptibility of the system to feedback in another manner. This results from the property of loud-

[*Continued on page 37*]

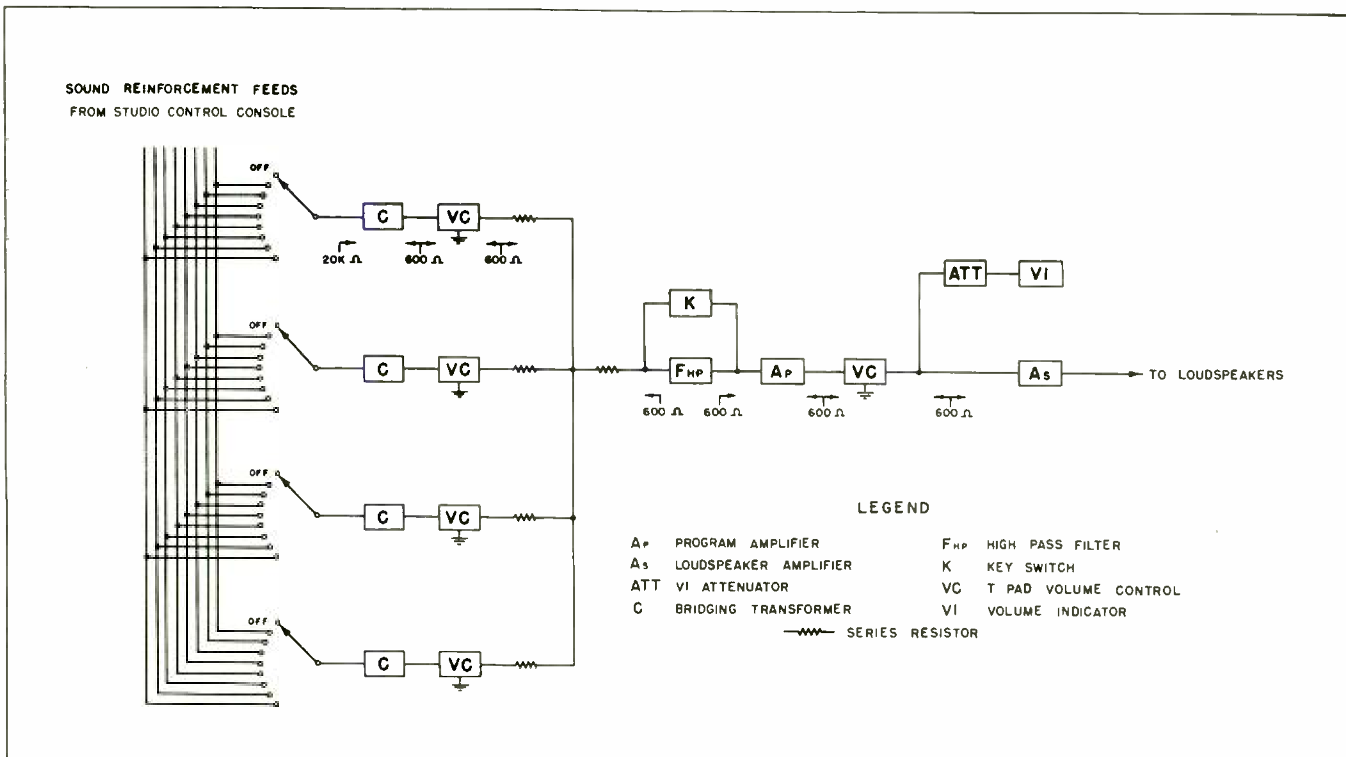


Fig. 4. Simplified block diagram of the circuit arrangement of the sound reinforcement mixing console of Fig. 3. Four mixer positions which may be bridged across any one of the eight program feeds are provided. The fact that the bridging transformers are incorporated in the sound reinforcement channels instead of the incoming program feeds results in a saving of transformers.

Noise Modulation in Recording

EMORY G. COOK

Causes of noise modulation in disc recording and how they may be overcome

DUE PARTLY to the influence of the f-m program and partly to the competition of various new forms of sound recording, standards of performance in the disc recording field are improving rapidly. The newer British phonograph records, together with the advent of feedback recording heads, improved disc materials, shapes of styli, etc., have all served to make possible the use of wider range reproducing playback facilities.

However, widening of frequency response and improving of all forms of distortion uncover from time to time certain residual "inherent" defects of systems, recording or otherwise. For example, the appearance not long ago of higher quality loudspeakers met with harsh words at first in some quarters because when they were attached to existing systems already in operation by the customer they drew aside the curtain on all forms of high frequency trouble—c r o s s-modulation, non-linearity, and even parasitics. The speaker itself was blamed. These defects had theretofore been buried in the much greater shortcomings of the previous loudspeaker or concealed by its limited frequency response, or both. In the same way, shortcomings of the established disc recording methods are gradually being exposed. One of the most glaring deficiencies in disc recording (which incidentally has a certain parallel in magnetic wire or tape) is the matter of noise modulation.

*Cook Labs, 139 Gordon Place, Floral Park, L. I., N. Y.

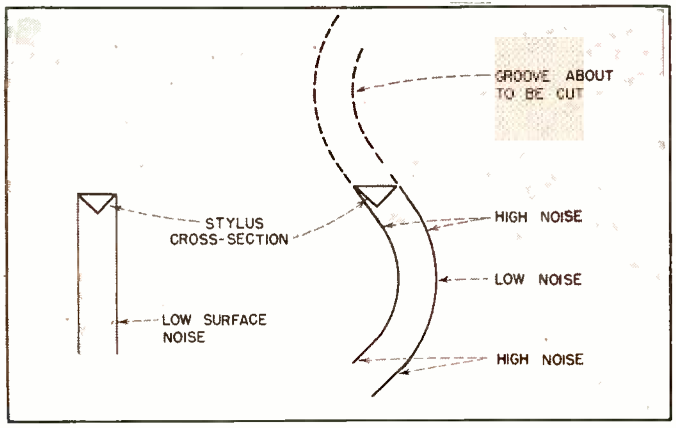
Introducing Noise Modulation

It is a matter of common knowledge with experienced disc recorders that one of the prime essentials to the obtaining of a quiet groove is the correct mounting in the cutter of the cutting stylus. The stylus set screw must of course come to rest directly upon the milled flat of the stylus shank in order that the stylus be pointing straight ahead. Otherwise, the stylus will be twisted in the mounting and a noisy cut will result. Yet a glance at *Fig. 2* will show that operation at an angle of twist is exactly what the stylus must do when engraving a signal in the groove. When the stylus is at the point in the sine wave shown at (*a*) it has no way of knowing—as far as noise is concerned—that it isn't being operated in a dead groove with twist (*b*).

The common method of measuring residual or surface noise in lacquer disc recording is to cut an unmodulated groove at the diameter and rpm indicated, insert in the playback circuit a high-pass filter to remove rumble and hum, and observe the resultant meter reading referred to "zero." As a practical matter the character of the sound thereby reproduced will be not unlike thermal agitation—random noise. But in this case the frequency spectrum of the noise is such that about 70% of the energy content lies between 3,000-10,000 cycles (if we were to double the speed to 150 rpm for instance, and use an appropriate reproducing head, the upper limit would naturally be extended). Unfortunately, this method of measurement is not necessarily a true index of performance of the stylus and lacquer, so far as surface noise is concerned, for in a way similar to that of intermodulation between two differing program frequencies, the actual reference noise level as measured by the dead groove method above, is modulating by the program itself. Unlike intermodulation, the noise modulation is *up*, i.e., when a cycle of program comes along, the noise in the sloping part of the groove is *higher* than it was in the unmodulated groove (see *Fig. 1*). The increase in noise may be in some way a function of the slope of the modulated groove (without regard for sign), but not necessarily so. It depends a lot on the stylus itself. The actual value of noise modulation is conveniently expressed in decibels, and is defined for the purpose of this discussion as being the number of decibels *increase* in noise obtained when a 35° slope occurs in the modulated track (see *Fig. 2*).

Listener Reaction

No mere butterfly chasing is being undertaken here in the matter of noise modulation. The writer is certainly *not* in the process of waving gleefully aloft a rare, microscopic, but otherwise utterly inconsequential specimen for examination by fellow entomologists. Any impulse on the part of the reader to suspect so must be reconciled with the fact that "before and after" listener tests made with regular styli and anti-noise modulation styli are hard to argue down. A "before and after" change so large in proportion that it is plainly distinguishable to the untrained ear stands out like a red nose in the measurements and instruments de-



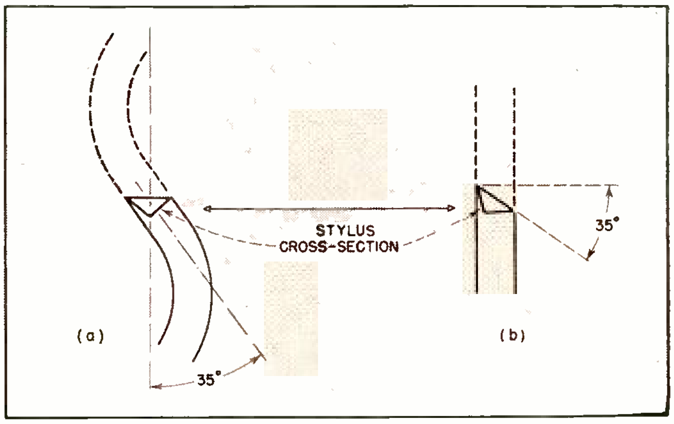


Fig. 1. The low noise-signal ratio measured in the dead groove (left) is only representative of what actually happens in one part of the a.c. alternation. Fig. 2 (right). The two methods of setting up to measure noise modulation—(a) dynamic, (b) static. At the instant shown, the stylus in (a) is just as embarrassed as the (b) stylus in its attempt to polish the sidewall to a noise-free surface.

partment, and can hardly be explained away any more easily.

It has already been well established in the art by numerous sources working in different ways, that cross or intermodulation is a more accurate index of listener reaction than amplitude or wave form distortion. It often does happen in reproducing sound systems that the phenomena which cause amplitude distortion also cause intermodulation at the same time; and since distortion content was at first easier to measure, this was used as a standard of performance.

Just as intermodulation between various program frequencies runs a close parallel to listener reaction in the overload level department, so noise modulation closely parallels reaction in the matter of *effective noise*. Although a conclusive probe into the matter of listener reaction has not been concluded, enough information is nevertheless at hand to warrant a few arbitrary statements concerned with the subject of the listening public vs. noise modulation.

1. When peak intermodulation is less than four per cent, frequency response reasonably flat to at least 8-10 kc and the whole playback channel including speaker is "clean," the effect of noise modulation is laid bare in its character as a residual trouble.
2. With any given set of circumstances (i.e., type of program, rpm, diameter, etc.) the use of high frequency preemphasis, since it produces sharper angles of groove excursion than a "flat" response, increases noise modulation, although it *decreases* the noise.
3. When shellac pressings originally recorded with no high frequency content over 5,000 cycles are played back into a channel flat to 15,000 cycles, the impression is given of distorted high frequencies. In listening tests it is easy to imagine that the recorded program contains components above 5,000 cycles even when such is known not to be the case. Modulation of the high shellac surface noise by the lower frequency signal produces this illusion. Indeed we do after a fashion have high frequency material present, in that the components of surface noise over 5,000 cycles heard in the wide-open playback system are being modulated or *cross-talked* at program frequencies.
4. The listener reaction ordinarily associated with plain *noise* is perhaps more correctly identified with *noise modulation*. There are some indications already that noise modulation, depending on a variety of circumstances, is the more important of the two factors.

For purposes of the present investigation an arbitrary stylus *Factor of Merit* has been devised which, although open for appropriate modification and change from time to time as the occasion demands, is nevertheless a signpost leading to an ultimate disposal of the problem.

$$\text{Factor of Merit} = \frac{\left(\begin{array}{c}\text{Number of db be-}\\ \text{tween conventional}\\ \text{noise and 10 cm/sec}\end{array}\right) - 40}{\text{Noise modulation (db)}}$$

In order to give an idea of the range of *Factor of Merit* values, figures in the neighborhood of 1.0 are acceptable, 2.0

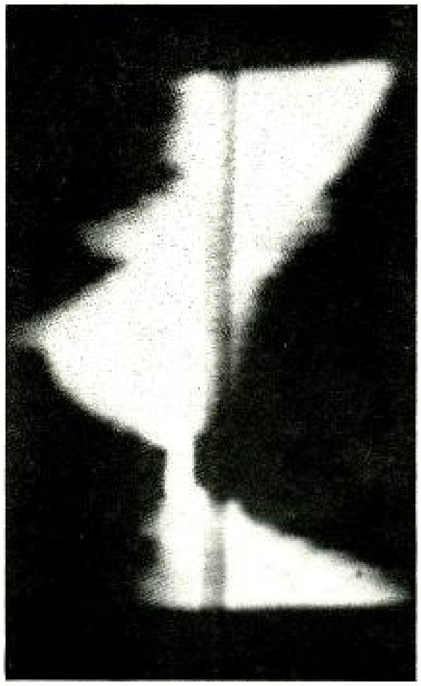


Fig. 3. Half-cycle envelope of noise modulation obtained from playing back tone groove. This is an exaggerated case, however, and Fig. 6 shows a more typical stylus.

is excellent, and 0.5 is very poor. Obviously, the relation is shaky and starts to break down for values of unmodulated groove noise level noisier than −45 db. Fortunately, it appears that it is somewhat easier to manufacture a cutting stylus with a good noise modulation figure if the unmodulated groove value demanded in the application is not too severe. It is much easier, for instance, to make a stylus with only 6 db of noise modulation if it is a −50 db stylus than if it is a −60 db stylus.

Measurement

The method of measurement used to establish the noise modulation figure for any given set of circumstances may be of two fundamental types, static and dynamic. The static measurement is easy to make, and as is so often the case with easy things, is not particularly reliable. It consists of turning the stylus 35° away on its own axis from the conventional position (*Fig. 2(b)*), cutting an unmodulated groove, and observing the increase in noise between this groove and an adjacent groove made with the same stylus without twist. The direction of twist must be tried both ways to correspond with the positive and negative velocities of the cycle in the groove. The unreliable feature of this measurement appears to be that since the stylus is operating under questionable conditions when twisted, the slightest imperfection in either it or the lacquer will hang up on the leading edge and stay there, causing a pessimistic reading, whereas in the dynamic method the stylus is continuously modulated or cycled as it would be with program, and particles which would be hung up under the static condition are thrown off. The dynamic method of measurement involves selecting a frequency which when operated to fill the groove to capacity will produce approximately 35° of slope as measured through the microscope. Naturally the frequency necessary to produce this slope will depend upon the diameter and the rpm, i.e., linear speed, of the groove. It is not surprising that the same stylus will show slightly different values of noise modulation at different linear speeds depending upon the dimensions of the various facets, etc.[1] The frequency necessary to produce the required angle will ordinarily be below 2,000 cycles so that in playing back the modulated (and noise modulated) groove

[1]Isabel L. Capps, "Recording Styli"—*Electronic Industries*, November 1946.

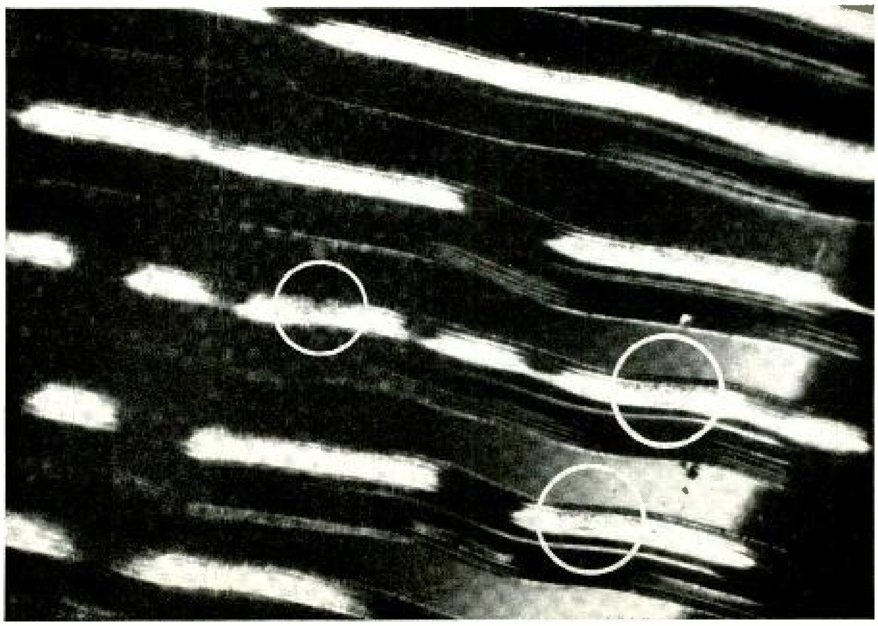


Fig. 4. Noise modulation in the sidewall is not hard to see with a low power microscope when the lighting is arranged properly.

Review of the Present Status of Magnetic Recording Theory

W. W. WETZEL*

PART II

In this series of three articles, Dr. Wetzel presents the first complete discussion of magnetic tape recording theory for engineers.

IN PART I of this article we discussed the hysteresis loop tester and how certain basic properties of magnetic materials could be obtained from readings on that instrument. We saw, as the result of demagnetizing forces, the residual induction in a recorded tape may be expected to decrease with decreased wavelength. It was further shown in a qualitative fashion that the coercivity of the magnetic material governs the remanent induction at short and the remanence governs the remanent induction at long wavelengths.

Part II is designed to familiarize the readers with the current status of erase, record and reproduction theory. Workers in the field are in general agreement on how a recording is erased or reproduced but there is disagreement on the mechanism operating during the process of recording with the aid of a-c bias. It is believed this disagreement is based on lack of appreciation of two factors: 1) the exceedingly complex magnetic history of a particle as it traverses the gap, and 2) the profound alteration of the induction picture which occurs when the demagnetization forces are brought into play upon removal of the tape from the recording head. The explanation offered here is based on two assumptions which are subject to criticism but which do have the merit of leading to results compatible with experiment.

Because it is a controversial matter more attention will be given to recording than to wipe and reproduction theory.

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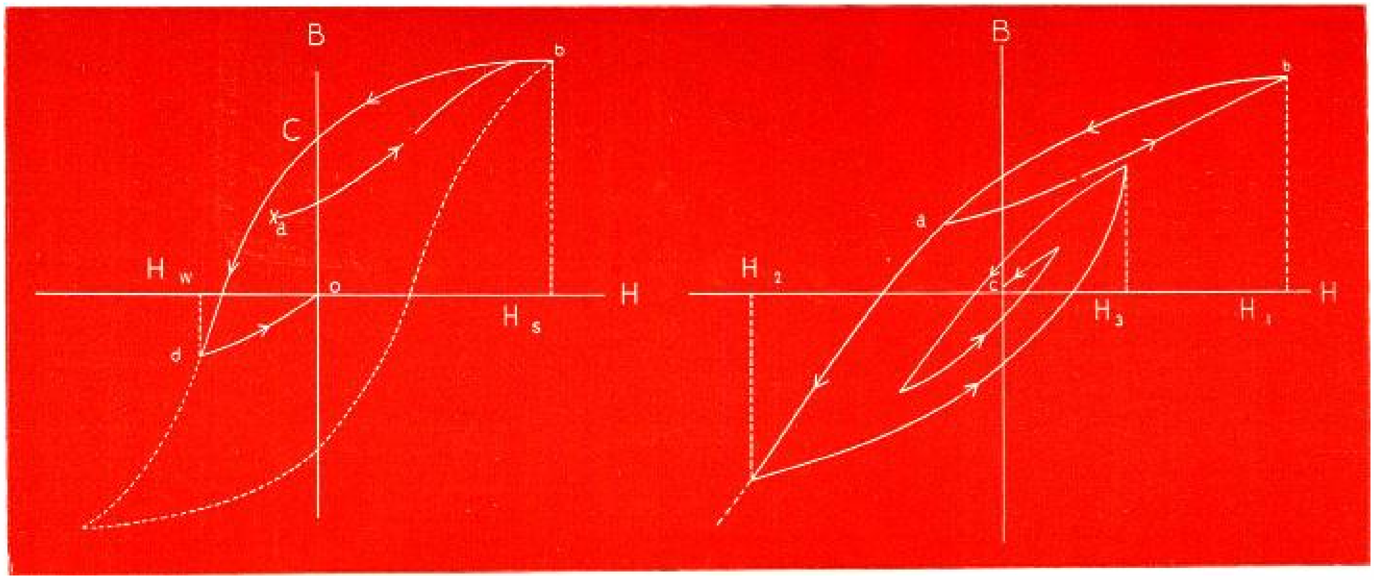
Erase

One of the outstanding advantages found in this form of recording is the fact that the records may be obliterated and re-used without deterioration of the medium. This process is known as magnetic wipe or erase. Obliteration of the record may be made with a d-c wipe in which the tape is brought to a magnetically saturated condition. The d-c wipe is required for systems employing d-c bias fields to obtain linear recording. In some cases, where the saturation field for the material is too high to permit the use of a-c erase, a d-c wipe with a permanent magnet is employed.

When a-c bias is used the recording medium should enter the recording head in as nearly as possible a neutral state of magnetization. This condition may be obtained by two methods, a-c wipe or a two step d-c wipe. *Fig. 1* shows the reversed d-c wipe accomplished by the application of two d-c fields. Starting from any state of magnetization "*a*" for the material, a constant or d-c field is applied by, say, a permanent magnet which is sufficiently strong to more than saturate the material. The application of this field brings the magnetic state to point "*b*." On removing the field the state moves along the saturated hysteresis loop to point "*c*." If following this a very precise negative field H_w is applied the state moves to point "*d*." Upon removal of this negative field the state changes along a minor loop to "*o*" or the non-magnetized condition. While it is theoretically possible to attain perfect wiping in this fashion, the precise value of H_w required is difficult to produce in practice. A further objection to this type of wipe is that the neutral condition is not stable. The noise level tends to increase with mechanical working of the tape after a wipe of this kind.

A better form of obliteration which

Fig. 1 (left). Showing the change in the state of a magnetic material during a d.c. erase. Fig. 2 (right). Diagramming the change in state accompanying an a.c. erase.



can result in noise levels below those of any usual amplifier may be attained by an a-c wipe. This form of erasure is stable in its final result and practical to apply. It is the form now generally employed.

It has been known for a long time that ferro-magnetic materials may be reduced to a neutral state if a series of decreasing fields alternating in sign are applied to the material. The process is illustrated in *Fig. 2*. If the magnetic state is originally that at "*a*," a positive field H_1 is applied which is just sufficient to bring the state to the apex "*b*" of the major loop on which "*a*" lies. This field is removed and a smaller negative field H_2 applied. Successively applied fields, decreasing in magnitude and alternating in

the value of H corresponding to the highest value of recorded induction. However, it is generally good practice to have several cycles of the wipe field saturate the tape. This guarantees that the wipe will remove overloaded recorded passages.

It is sometimes said that a high coercive force material is difficult to wipe. This is not in agreement with the best theory of erase. It should be stated as "a material with a high saturation field (H_s not H_c) is difficult to wipe."

Figure 3a shows the flux distribution in the neighborhood of an erase head gap. A well designed erase head should saturate at the pole tips, increasing the leakage flux from points remote from the gap. The values of H_h and H_v in the neighbor-

portion of the input-output characteristic, the magnetic fields used to produce this effect in recording have come to be known as bias fields. The word bias is an unfortunate choice in this application since its use tends to influence our thinking in terms of vacuum tube theory rather than in magnetic terms. This has resulted in erroneous applications of the transfer characteristic curve.

Three types of bias have been proposed and used. D-C bias is employed on a previously saturated recording medium to bring the operating point to a position on the saturated hysteresis loop from which it will return to $B_r = O$ upon removal of the field. This is precisely the procedure used in d-c erase, *Fig. 1*. An audio field superimposed on the d-c bias field will leave on the tape values of B_r ...or below zero which are proportional ... instantaneous value of the audio ... Many discussions of this method ...ding have been published. [1,2,3] ... the most detailed ... has reviewed.

... recording employs ... tape to which an ...xed with the audio ...s resulted in alter...ted and less than ... when the tape is ...cording head, de...nd to average the ...ng less than the ...his type of record...udio induction cor... longitudinal mag... zero. As will be ... gives rise to back-

...gnetic Sound Re... and Ringheads," ... 20, 1937.

...ound Recording on ... *System Technical*

...etic Recording and ...ns in the Broadcast ... 423, August 1941.

...different wavelengths. ...alculated from Fig 5.,

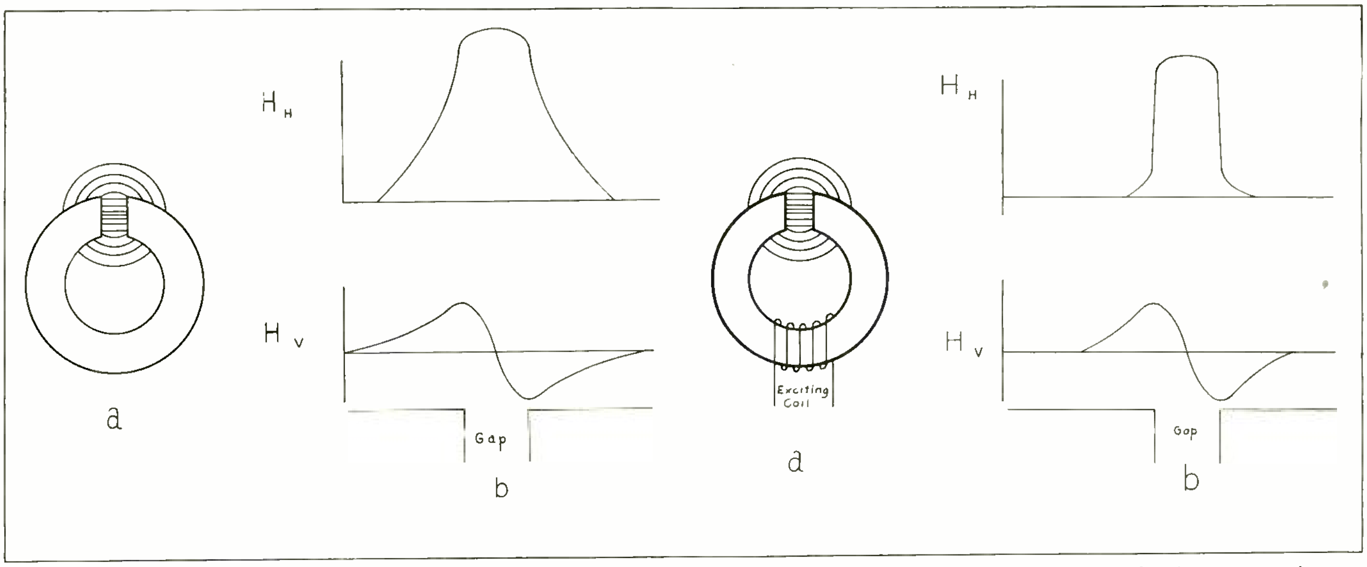


Fig. 3 (left). Sketch showing the field distribution near the gap of an erase head. Fig. 4 (right). Sketch showing the field distribution near the gap of a recording head.

ground noise. No paper on the theory of a-c bias applied to a saturated tape has come to the author's attention.

The third method which employs a magnetically neutral tape with a mixed audio signal field and ultrasonic bias field has been almost universally adopted since it results in a greater undistorted signal to noise ratio than either of the above techniques. Unfortunately no detailed account of the theory has appeared which is completely satisfying. Toomin and Wildfeuer[4] have made studies of the behavior of minor hysteresis loops superimposed on major loops and proposes an explanation based on their findings for linear transfer characteristics in recording. Holmes and Clark[5] point out that certainly the audio fields and in many cases the bias fields have not been established for the number of cycles required to produce stabilized major and minor loops. They propose an explanation in which the state of magnetization of the medium is traced as the material passes across the gap and find it to have an induction dependent to a large extent of the maximum field encountered during the transit of the gap.

This explanation is the most satisfying published. One apparent objection is that the same remanent flux is found in many successive particles under certain conditions and remanence is therefore not a smoothly varying function of position along the tape.* It will be shown that because of the profound alteration of the flux distribution during demagnetization many such irregularities in the initial flux distribution will be smoothed out.

[4]H. Toomin and D. Wildfeuer, "The Mechanism of Supersonic Frequencies as Applied to Magnetic Recording," *Proc. I. R. E.*, p. 664, November 1944.

[5]L. C. Holmes and D. L. Clark, "Supersonic Bias for Magnetic Recording," *Electronics*, July 1945.

To appreciate the complexity of the history of a magnetic particle as it passes over the recording head suppose we refer to *Fig. 4*. Sketch "*a*" represents a recording head with its gap and the associated field fringing out from the gap. It is this fringing field which is effective in producing the magnetization of the tape as the tape passes, say, from left to right across the top of and in contact with the head. The exciting coil carries a current which is the sum of an audio and an ultrasonic component. At any instant the field on some one plane over the gap is shown in *Fig. 4b* broken into horizontal and vertical components. For short distances above the head the value of the field varies approximately as the inverse first power of the distance of this reference plane from the head. Superimposing the time variations of the field, the space variations of the field, the two components of the field to be considered and the complex behavior of a ferromagnetic material in two dimensions, we begin to appreciate the difficulty of tracing the particle's magnetic history.

In order to make progress in the explanation let us assume that as a particle traverses the gap, some process leaves a horizontal component or remanence which is proportional to the instantaneous value of the horizontal component of the field at some point in the gap. If this is true, we are able to explain the action of recording with a-c bias. It will be shown, in fact, that this hypothesis is too detailed and that it is not necessary to have a remanent flux exactly proportional to the instantaneous value of the field in order to obtain linear output.

*For those interested in the details of this argument see *Fig. 4*, Reference *5*. All particles entering the gap between t' and P should leave the gap with very nearly the same horizontal component of remanent flux. This is true since the quarter cycle of the field following S will only extend the portion of the minor loop R, S. For all values of the quarter cycle above zero the remanence should be the same. For the remainder of the quarter cycle the remanence will be only slightly less.

Before starting the explanation let us show how we obtain a curve which Holmes and Clark call the transfer characteristic. If the loop tracer is called into use, we may examine the major hysteresis loops for a given tape for certain maximum values of the 60-cycle field. If a family of curves is observed for values of H_{max} between zero and saturation, we have information which allows us to plot the B, H curves shown in *Fig. 5*. From these curves, of which only two are shown, we read the remanent induction in the tape to be expected when the field goes to zero from each value of H_{max}. This gives data from which to plot the transfer characteristic curve shown as Curve 1 in *Fig. 6*.

Line aa', *Fig. 5*, represents a calculated demagnetization function for a certain audio wave length on a tape. Its intersection with the hysteresis loops defines the maximum value of induction which a tape will support at a given wavelength for given impressed fields H_{max}. The value of B_r at the intersections may be plotted against H_{max} to obtain Curve 2. The audio signals applied to a tape may be expected to transfer according to Curve 1 for very long wavelengths and according to curves of Type 2 for shorter wavelengths. While this is correct for d-c bias, under the influence of a-c bias this is not true since the demagnetizing forces involved are a combination of those for the a-c bias wavelength and the audio wavelength. The actual final transfer will be shown to deviate markedly from such curves.

Let us make the second assumption required for the explanation and suppose that for long wavelengths the transfer characteristic curve before demagnetization is Curve 1 of *Fig. 6*. Since Curve 1

was established on the basis of stabilized hysteresis loops, this is not strictly correct. The true transfer curve must be one similar in shape but with somewhat lower B_r values.

Suppose we borrow the type of illustration used in obtaining input-output curves for radio tubes. Let us plot the input ultrasonic field against time on the vertical axis and in the usual fashion trace the output induction as a function of time on the horizontal axis. This approach is illustrated in *Fig. 7*. So long as the tape is in contact with the recording head, the dashed output curve represents the horizontal component of induction. Upon removal from the head, poles will form on the tape and the associated demagnetizing forces will reduce the induction to a value indicated by the solid output curve. The remanent induction on the tape will be that which would have resulted had we used a transfer characteristic derived from the intersection of some line *b-b′* with the hysteresis loops of *Fig. 5*. The slope of *b-b′* is presumed to be calculated from the demagnetization constant for the bias wavelength. With no audio signal on the recording head and the bias field adjusted to reach the neighborhood of the limit of the linear portion of the transfer characteristic, it is seen that only a very low amplitude induction of ultrasonic frequency is impressed on the tape.

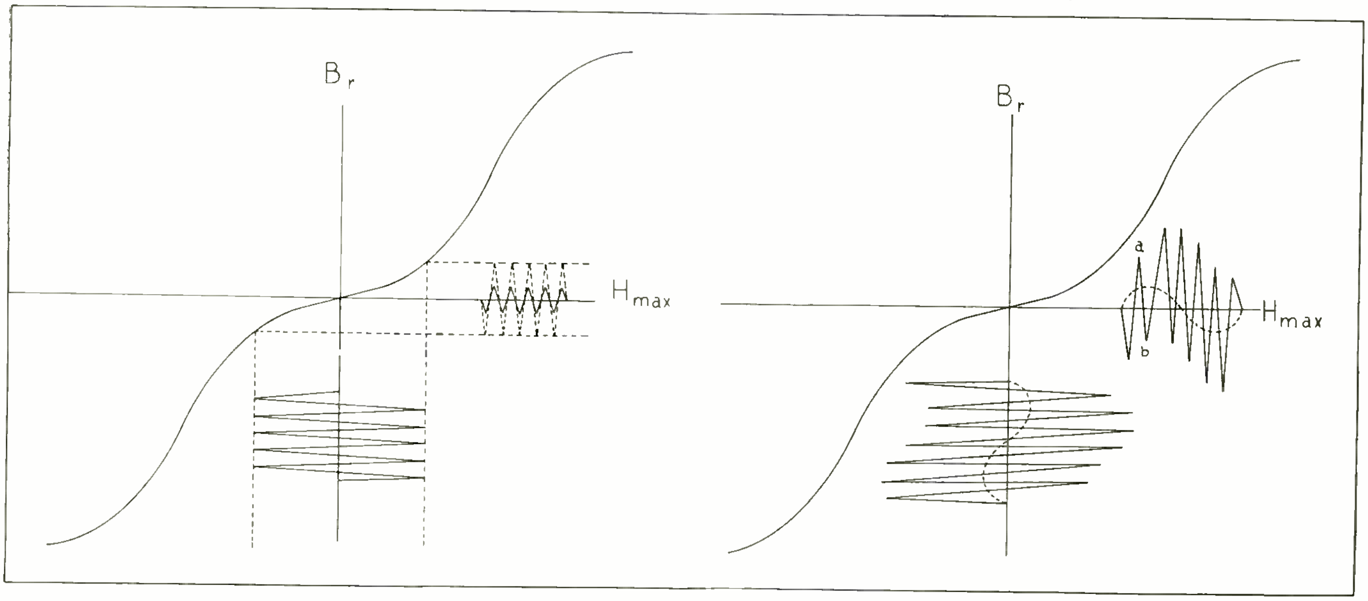
In recording with the use of a-c bias, the bias current and audio current are added in the coil of the recording head. This results in the applications of fields to the tape which are proportional to the sum of the instantaneous values of the bias and audio currents. The wave form of such an "input" field is illustrated in *Fig. 8*. (It is noted that the bias field is not modulated by the audio-frequency field as a carrier wave is modulated in radio transmission.) We assume that, before removal from the recording head, the horizontal component of induction in the tape is proportional to the instantaneous value of the field at some point in the gas and that the transfer characteristic is a curve of Type 1, *Fig. 6*. We proceed in the conventional manner to trace the output induction shown as the solid curve on the H axis of *Fig. 8*. This solid curve shows the condition of induction to be expected in a tape which has passed the gas but has not been subjected to demagnetizing forces by removal from the head.

The question of what happens upon moving the magnetized tape past the head may be answered qualitatively by considering the two sets of demagnetizing forces introduced. One set of forces is attributable to the poles associated with the bias frequency distribution of flux and the second set is connected with the average of the bias wavelength poles and may be attributed to the audio wavelength.

As a starting point in the explanation suppose we consider one bias frequency oscillation of the recorded induction *a*, *b* (*Fig. 8*) for a very long wavelength of audio signal. The oscillation *a*, *b*, which occurs at the point of maximum induction for the audio output and therefore at the point of zero demagnetizing force for the long wavelength, may be considered as the sum of a d-c component and a sinusoidal component of induction. The only demagnetizing force to be considered is that of the poles associated with the sinusoidal component. The equilibrium which will be established between induction and poles for the sinusoidal component will be that expected from the transfer characteristic curve constructed for the bias frequency. This requires that the over-all induction change to the picture of a very small oscillation about the zero value of the induction for each of the original sinusoidal components. The final picture of the output induction is shown as a dashed line drawn through the zeros of each bias frequency oscillation in the output curve of *Fig. 8* which, for simplicity, does not show the small superimposed oscillations at bias frequency. It is seen that the effect of the a-c bias superimposed on the audio signal is to impress on the tape a reproduction of the audio signal in terms of induction. The effect of the bias is to remove the first non-linear portion of the transfer characteristic curve.

Although the small bias frequency oscillations superimposed on the audio are unimportant practically, since they are subject to further smoothing action in the reproduction system and at any rate would never be heard, their existence can be used to substantiate in part the assumptions on which we have proceeded. If the output of a tape on which a low audio frequency has been recorded is examined in an oscilloscope, the bias frequency oscillations may be seen. Since the output of a pickup is the derivative of the induction in the tape, the bias oscillations of greatest amplitude are found near the zeros of the output wave and disappear near the peaks. This is exactly the position in which modulation noise is expected to reach a maximum value, so the picture in the scope is sometimes attributed to that form of noise. The sweep frequency is adjusted to the audio value and the irregular position of the bias frequency pips does resemble the picture of random noise. A second experimental check may be had by recording a low-frequency tone on a tape running at high speed. If this tape is played back at a considerably lower speed and the output examined on an audio wave analyzer, the bias frequency appears in the output at the expected

Fig. 7 (left). Showing the use of the transfer characteristic in obtaining the output induction for the bias field. Fig. 8 (right). Showing the combined audio and bias input fields and how they are transferred to induction in the tape.



new frequency with an amplitude which is larger the greater the value of the recorded audio amplitude. This increase in amplitude is what would be expected if the demagnetization of the bias component of induction takes place according to the transfer characteristic curve for the bias frequency.

We have seen that at low frequencies where demagnetizing forces of the audio pole distribution can be neglected the transfer characteristic curve does not predict the correct output directly. A transfer characteristic having something less than half the B_r values of Curve 1, *Fig. 6*, must be used if the conventional transfer procedure is employed where the peaks of the input field are used to trace the output induction.

For short audio wavelengths we must take into account the demagnetizing force due to the pole distribution associated with the audio wavelength. This results in a further reduction in the amplitude of the remanent flux. Two general procedures may be followed in obtaining the final answer. We may either calculate the transfer on the basis of Curve 1, *Fig. 6* and apply successively the demagnetization calculations for the bias, then the audio wavelength, or we may start with a transfer characteristic Curve 2, *Fig. 6*, which contains the demagnetization effect for the audio wavelength and calculate the induction from it in the same manner as was used for long wavelengths. These two procedures lead to essentially the same results for the audio frequency induction. The first method results in a bias ripple amplitude which is independent of the audio wavelength. The second method would lead to a ripply amplitude which decreases in audio wavelength. An experimental test of the amplitude of the bias frequency output as a function of the audio frequency recorded tends to support the first method of correcting for the two demagnetizing forces. Theoretically the first method of correction is the more satisfactory.

We conclude that, while the simple picture of demagnetization outlined in Part I applies directly to recording with d-c bias, the picture in the case of a-c bias, used either with d-c or a-c wipe, is complicated by the introduction of the bias demagnetization forces. The fact that strong demagnetizing forces are associated with the bias frequency, and that the remanent induction undergoes profound alteration under their influence, eases the requirements of our first assumption. It is seen to be unnecessary that the mechanism assumed in the gap produces a smoothly varying remanent induction before demagnetization. What is required is that the mean value of the induction over one cycle of the bias frequency be proportional to the value of the audio field.

Reproduction

As we have seen, a tape after leaving the recording head carries with it a value of remanent induction which from point to point is proportional to the audio signal impressed on it. After leaving the head, flux lines threading the tape establish return paths through the air which are maintained until the tape is brought into the neighborhood of other permeable material or the tape flux is altered by erasure.

In *Fig. 9a*, a tape is shown on which a sine wave has been recorded. Lines of induction in the tape and return flux through the air are shown schematically. No attempt has been made to illustrate the small flux circuits which represent the recorded bias frequency since these are not important in reproduction theory. *Fig. 9b* shows the redistribution of flux which occurs for one position of the playback head in contact with the tape. Flux which normally found a return path through the air is redistributed through the low permeability core of the reproducing head and threads the pickup coil. A voltage will be developed in the pickup coil which is proportional to the time rate of change of the flux threading the coil. For wavelengths long compared with the gap width, the amount of flux by-passed through the coil is essentially equal to the induction crossing a plane, such as *a* or *b*, *Fig. 9a*, cutting the tape at the center of the gap. As the tape moves across the head, it is easily seen that the output voltage developed is proportional to the rate of change of induction in the tape under the gap.

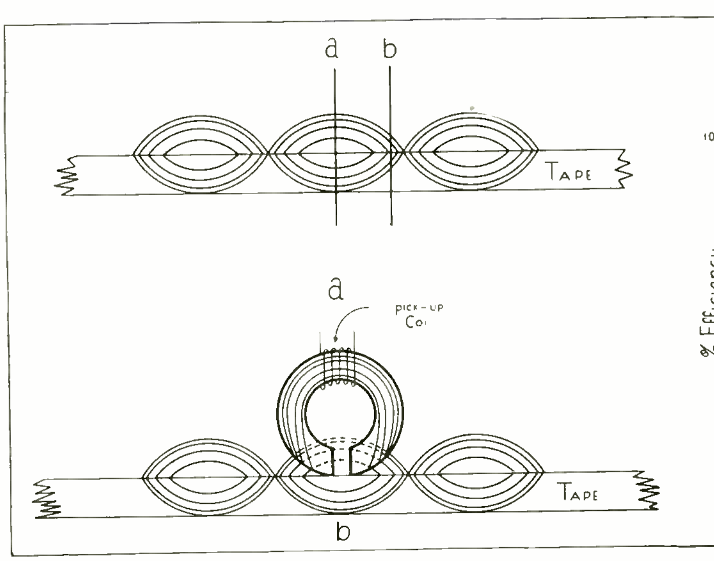
An interesting effect, which results in periodic reduction of the reproducing head output, is seen if one examines the flux distribution as the wavelength becomes increasingly shorter with respect to the gap width.

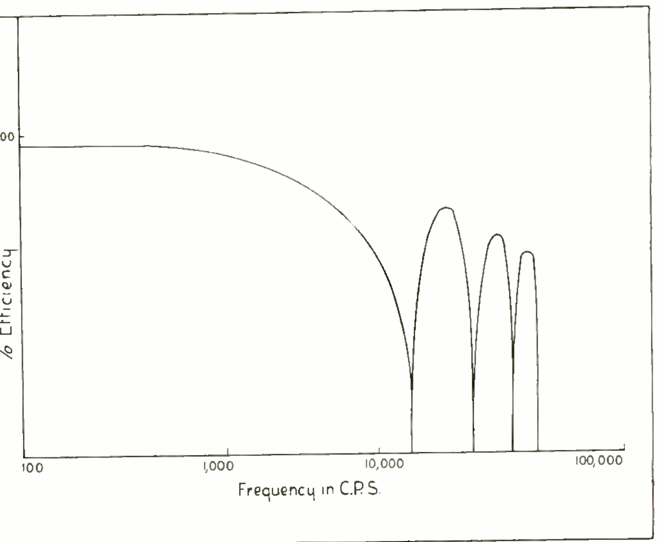
It may be shown that with decreasing wavelengths some flux lines find the air path to be that of least reluctance. These lines do not thread the coil and represent a voltage loss in the pickup. This loss increases up to the point where the wavelength equals the gap width where the output of the head falls to zero. The fall to zero output occurs at each frequency for which the gap width is an integral number of wavelengths, with partial recovery of pickup efficiency for intermediate wavelengths. A qualitative plot of the efficiency of the reproduce head as a function of frequency is shown in *Fig. 10*. The decrease in efficiency for successive peaks of the curve is due to the increase in the ratio of the core reluctance to that of the decreasing air paths' lengths, as the wavelength shortens.

This short discussion of the effects of gap width suggests that a decrease in gap opening tends to increase the high-frequency output. This is true up to the

[*Continued on page 37*]

Fig. 9 (left). Illustrating flux lines in the tape and air associated with a recording. The effect of placing a high permeability head on the tape is to rearrange the air flux so that it threads the pick-up coil. Fig. 10 (right). Showing the efficiency of a reproducing head as a function of frequency.





Two-Way Speaker System

C. G. McPROUD*

PART II

The second of three articles describing the design and construction of an excellent two-way speaker system.

IN THE previous article of this series, instructions for making the high-frequency horn were presented, together with a discussion of the elements of a two-way system and the choice of a speaker unit suitable for use with a system designed for a crossover frequency of 900 cps. Since it is obvious that a two-way system requires two separate speakers, properly coordinated into a single unit and working as one sound source, the low-frequency section must also be provided in order for the complete system to be capable of covering the entire audio range. It is the purpose of this article to describe a suitable low-frequency speaker housing, together with the choice of the speaker itself.

It will be remembered from the first article that the two-way loudspeaker consists of a high-frequency unit and horn, a low-frequency speaker and a suitable enclosure for it, and a dividing network to channel the correct frequencies to the two sections of the system. There is nothing new about the system being described—similar arrangements have been on the market for years. The original theatre systems were single-unit devices, with a diaphragm-type driver being attached to a long exponential horn. These speakers were fairly satisfactory, considering the remainder of the sound equipment. However, with continual improvement of the recording and reproducing equipment, the need was felt for better loudspeakers, and the wide-range system made its appearance. Without tracing the evolution of speakers, let it be said that the present high-quality speaker systems are almost without exception dual-unit devices. Since these systems are so capable throughout the entire frequency range, smaller systems employing a similar design have also become popular for better reproduction in the home and in recording and broadcast monitor rooms. The smaller systems were scaled down in physical size, at a sacrifice of power handling capacity and the lower frequency register. However, they are still the most capable systems available, and the main deterrent to their more wide acceptance has been the cost and the space required.

While the space requirements cannot be appreciably reduced, the cost is subject to considerable reduction provided the equipment may be assembled, without too much trouble, by the ultimate user. Obviously, both the high-frequency unit and the low-frequency speaker should be purchased from a reputable manufacturer, since it would be folly to try to make them. However, the high-frequency horn can be constructed without too much special equipment, and the low-frequency baffle can be built by anyone who can handle woodworking tools with any degree of proficiency.

The Low-Frequency Section

The low-frequency section of a two-way system may consist of a folded horn, a plain baffle, an infinite baffle, some form of acoustic labyrinth, or a reflexed enclosure. While the best performance can usually be obtained by the use of some form of folded horn, there is some question as to whether the response between about 400 and 900 cps is satisfactory, since the upper frequencies passed by the low-frequency speaker are required to traverse the turns in the horn, and some loss may be expected in that region.

The flat baffle is the simplest arrangement, but it is deficient in the very low register, even if it becomes of large dimensions. The infinite baffle, such as the wall of a room with the speaker itself being in another room, or in an attic, is reasonably satisfactory, and is preferred to the plain baffle.

The acoustic labyrinth type of speaker requires a lot of complicated design and construction, and it is doubtful if it is any better than a cabinet form of mounting, provided some means is taken to reduce the effect of cone resonance. Some speakers are suited for mounting in a completely enclosed cabinet of smaller dimensions than normal, with the stiffness of the enclosed air serving as the damping.

For over-all use, considering the space required, the radiation efficiency of the speaker at low frequencies and the external appearance—which must be considered unless the user is a bachelor—make the reflexed cabinet a logical choice. It should not be assumed that this is the ideal speaker housing for the reproduction of low frequencies, but considering the compromises that must be made occasionally, it has certain advantages.

For these reasons, the low-frequency speaker selected for use with the two-way system described here is of the reflexed type, of a size consistent with the speaker selected.

Reflexed Cabinet Design

The design of a suitable reflexed cabinet requires the possession of considerably more information than is usually available to the constructor, besides requiring a knowledge of the basic principles of acoustics and the methods of calculating acoustic compliance, mass, and a variety of other characteristics. Among this data that must be at hand is the resonance frequency of the loudspeaker cone, and in only a very few instances is this figure available from the manufacturer without considerable trouble.

On account of this, a much simpler empirical method of determining the

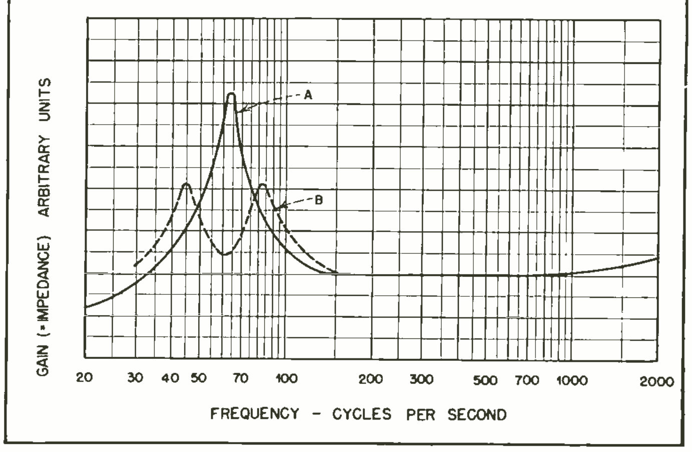


Fig. 1. Curves indicating speaker impedance obtained by measurement of gain of an amplifier with pentode or beam-power output tubes. A represents curve of speaker mounted in open back cabinet, showing effect of cone resonance. B represents curve of speaker mounted in reflex cabinet with port adjusted so that both peaks are of equal amplitude.

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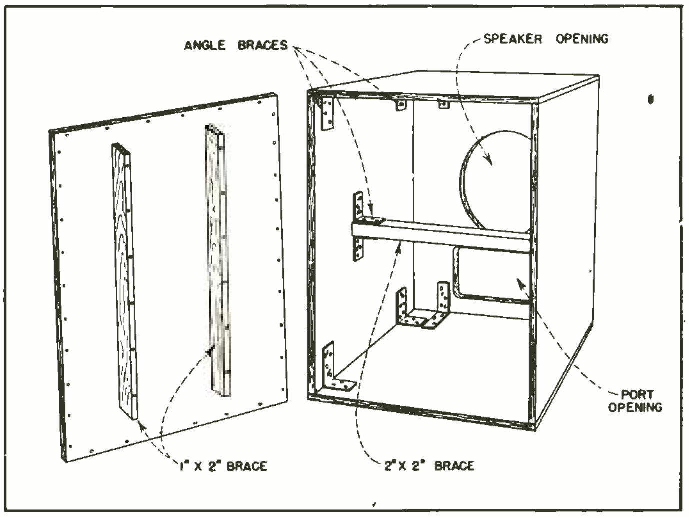
size of a cabinet is suggested. This method is based on the physical size of the speaker, and while it may be far from the classical method of designing a box, it is a slight improvement over one method employed in the design of an early commercial speaker enclosure. The engineer in charge of this design admitted that the box was as big as he could make it and still get it into the back of his sedan.

Unfortunately, this particular box was not sufficiently large, and the over-all result of the device was to introduce a peak in the response at about 70 cps, but in spite of that, the speaker was better than any others then available.

The empirical method is based on the following figures: Use a box which has a volume in cubic feet which is numerically equal to the nominal radius of the speaker in inches. Thus, using this method, a 15-inch speaker requires a cabinet volume of 7.5 cubic feet. This is not too far from a reasonable proportion, as has been determined by experiment. The port area is claimed by good designers to be equal to the actual radiating area of the cone, which is somewhat less than that calculated from the nominal diameter of the speaker. For example, a certain 15-inch speaker requires a 13½-inch hole for mounting, but the actual diameter of the conical section of the cone itself is only 12½ inches. Thus the suggested port area should be 123 square inches. Experiments have indicated that this is somewhat too large for the average speaker and enclosure, so resorting to an empirical formula again, the port opening may be taken as 0.8 times the area of the loudspeaker opening, to allow for adjustment to the optimum size.

Fig. 2. Internal bracing of heavy plywood cabinet used for low-frequency speaker. All joints should be glued, and assembly should be made with woodscrews rather than nails. Back must be thoroughly braced, and should have adequate number of screws to hold it securely to prevent vibration.



While the design of a suitable box may be done by means of a number of complicated formulas, the final result—as with anything in the audio field—is how it sounds when completed. Therefore, having decided upon a reasonable port size, it is much simpler to make the box and then experiment with the output until the result is satisfactory. This may be done by simple experimentation with the box, an amplifier, and an audio oscillator, in a method to be described later.

The reflex cabinet serves principally to increase the radiation of a loudspeaker at frequencies below those where it normally loses its efficiency. A secondary reason for the box is to alleviate the effects of cone resonance, which normally occurs at a frequency where it is objectionable. If the increased impedance at resonance can be corrected by the design of the cabinet, the response is sure to be better in that region.

Resonance Effects

The effect of speaker resonance manifests itself as an increase in impedance at the resonant frequency. While this in itself is not particularly important if triodes are used in the output stage, it makes a considerable difference when pentodes or beam tubes are used, unless there is a sufficient amount of inverse feedback to stablize the output impedance of the amplifier. If a curve of the impedance of a speaker is made with an open-back cabinet, it will be found that it resembles that shown in *Fig. 1*. Such a curve can be made by measuring the gain of a beam-power or pentode amplifier with the speaker as the termination, and covering the frequency range from about 30 cps to at least 300 cps. The impedance of the speaker affects the gain of the output stage, and consequently the gain-frequency curve represents the variations in impedance.

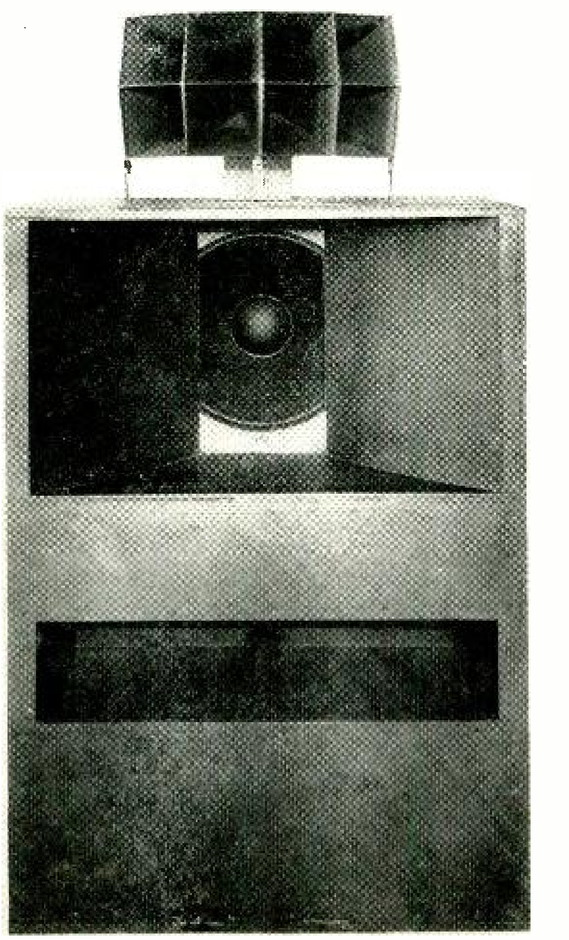


Fig. 3. Typical commercial form of two-way speaker system employing separate high- and low-frequency speakers. Construction of horn similar to these was described in article in the November issue.

When the back is put on a ported cabinet, the gain-frequency curve exhibits two peaks, one above the resonant frequency, and one below. Optimum results are obtained when the two peaks are of equal amplitude, as shown at (*B*) in *Fig. 1*. The relative amplitude of the two peaks can be varied by changing the size of the port, and the port opening should be so arranged that the two peaks are equal.

With this in mind, it is seen that the entire speaker enclosure may be constructed with a number of empirical formulas, and then by changing the size of the port to obtain the correct response curve, the optimum performance can be obtained. From a construction standpoint, it is advisable to make the port somewhat larger than the anticipated final size, so that the effective opening can be reduced by the use of an additional piece of wood mounted inside the cabinet.

Any *good* 12 - or 15-inch speaker may be used with this system. In order to simplify the dividing network, it is necessary that the impedance of the low-frequency speaker be 8 ohms, since that is the impedance of the high-frequency unit selected for use with this system. While it is possible to design a dividing network which will feed two speakers of differing impedances, it is much simpler to select a low-frequency speaker of the correct impedance in the first place.

Suitable speakers for this purpose are Jensen P12-N and P15-N in 12- and 15-inch sizes, or the PMJ-18 in the 18-inch size, although the latter is quite expensive. The General Electric Type 1201C is an excellent 12-inch speaker and Cinaudagraph 12- and 15-inch speakers of the public address type are suitable. Another extremely efficient speaker for this purpose is the Altec-Lansing Model 803, which is used in their Model 800 two-way system. However, any of the speakers listed will give satisfactory results, provided the enclosure is designed to work with them.

Several manufacturers also make suit-

able reflex enclosures, and one of these may be used instead of going to the trouble of making the housing. However, in order to obtain the best results, the housing selected should be one which is intended for use with the particular speaker chosen for the system.

Cabinet Construction

The actual contruction of the cabinet presents no particular problems. Considering, for example, an enclosure for a 15-inch speaker, it is desired to have a volume of approximately 7.5 cubic feet. The cabinet proportions required to make an over-all structure which is pleasing to the eye are such that the height must be approximately 1.5 to 1.6 times the width. Since the upper section must contain the high-frequency horn and unit, the top 10 inches of the entire speaker must be deducted from the total height in calculating the possible size of the enclosure itself. Thus, assuming a 7.5-cubic foot box the net proportions of the enclosure may well be approximately 1:2:2½, and from this proportionality the actual dimensions of the box can be determined. Allowing a wall thickness of one inch, the outside of the box for a 15-inch speaker arrives at a size which is approximately 16 x 30 x 35.

The construction should be as solid as it is possible to make it within practical limits. Three-quarter or seven-eighth inch five-ply is suitable, and while somewhat difficult to obtain under present conditions, is recommended for the enclosure. If the plywood panels are assembled with glue and wood screws and braced with steel angle brackets on the edges of sides and front, it will be sufficiently sturdy. It is a good plan to install a 2 x 2 brace through the center of the box from side to side, making it detachable so the speaker may be mounted. Such a detachable brace, however, must be solid when in place, and heavy bolts should be used for attaching it.

The back cover for the box should be arranged to be attached with heavy wood screws, and should preferably be braced with two pieces of 1 x 2 mounted edgewise, running from top to bottom. *Figure 2* shows the suggested bracing for the entire cabinet.

After the box is completed, it should be given a coat of shellac on the inside, and then lined with a sound absorbent material. Rock wool blanketing of 2-inch thickness is especially desirable for this purpose, but if it is not readily obtainable, the box may be lined with two thicknesses of Ozite, the padding material used under rugs. This material should be glued to the cabinet, and tacked with large-headed nails at six-inch intervals. Sufficient clearance should be left around the speaker opening to permit its installation directly against the front of the cabinet, and since the port opening is not yet determined finally, it is suggested that the space between the bottom of the port and the bottom of the front panel be left uncovered.

Before mounting the speaker, a circle of expanded metal should be obtained for installation in front of it as a protection against damage to the cone. The speaker is best mounted with "T" nuts on the outside, using 10-24 screws through the speaker frame and the front of the cabinet. This type of mounting permits the removal of the speaker, whenever desired, with the greatest ease. When the speaker is in place, with a suitable lead extending through the port, the back may be screwed on.

Adjusting the Port

Assuming that the port has been cut in rectangular shape with an area of approximately 0.8 times the speaker opening, the speaker is connected to an amplifier and a gain-frequency curve made. Assuming also that the amplifier is flat

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Notes on Wide-Range Reproduction

J. N. A. HAWKINS*

THERE HAS BEEN some controversy for many years over the desirability of high fidelity in radio and record reproduction. Various tests have been made on large and small groups of listeners, and a wide variety of conclusions have been drawn from the data obtained. It appears that much of the apparent discrepancy between the divergent conclusions can be explained by the fact that most of the investigators have overlooked a factor in the tests. This factor is of the utmost importance and might be termed "Audience Attitude," or "Audience Objective." Why are they listening, what do they want to hear, and what do they not want to hear, at different times?

Cidental Listening

This complex matter of "Audience Attitude" can usually be simplified by dividing all audiences into two groups: "Incidental Listeners," and, creating a new word to imply the positive form of the negative *incidental*, "Cidental Listeners." Cidental listening might be described as that kind of listening which centers all mental attention strictly on the program material. All distractions are extraneous and are discouraged. Incidental listening is quite different from cidental listening. The incidental listener is primarily doing something else, rather than listening to the program material. The music, for example, provides only a background for some other activity and mental attention is focused away from the program itself.

At this point it might be well to determine a suitable definition for "High Fidelity." It is suggested that an ideal definition might be that high-fidelity program material is that which cannot be told from the original. A useful working definition must be commercially realizable and thus may be something less than absolute perfection. This working definition can be expected to change as the art progresses, but, for purposes of discussion, a very good 1947 system might be described as follows: With microphone to horn gain about zero (acoustically), the transient and steady-state frequency responses should be within 5 db of each other, and neither curve should depart more than 10 db from the 1,000-cps transmission over the range from 80 to 8,000 cps. Over this same region the overall reverberation time should not vary by a factor larger than three. Background noise in the absence of program material should be 50 db below maximum desired level; noise in the presence of program material may rise to 30 db below program material on loud passages without being objectionable. At maximum level the 60/1,000 cps intermodulation distortion may reach 30 per cent, *provided* the fifth-harmonic distortion, (400/2,000 cps), does not exceed 0.3 per cent. At all output levels more than 10 db below maximum the intermodulation distortion should be below 10 per cent and the fifth-harmonic distortion below 0.1 per cent.

It is probable that many will consider the system described above as only medium fidelity. The answer to that objection is that probably fewer than 100 homes in the world have anything any better, no advertised radio receivers can possibly equal it, and few broadcast station monitors even approach it. It could be realized today by adding about $20 to the cost and $100 to the price of the average $200 FM console receiver.

Therefore, with a definition of high fidelity at hand, debatable or not, let us

[*Continued on page 35*]

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Impedance Matching

O. L. ANGEVINE, JR.*

First of a series for beginners in the sound engineering field.

THE TERM "impedance matching" is one of the most misused expressions in the technical literature. For example, as applied to the design of a sound distribution system, the term is used to signify matching (*a*) for maximum power, (*b*) for lowest distortion, (*c*) for a desired division of power between loudspeakers, and (*d*) for the proper termination of a transmission line to prevent echoes or reflections. For each of these applications, the word "matching" implies something different, but often the term is used as if it were the same thing. This discussion of impedance matching distinguishes between these various meanings and considers each separately. Examples of situations frequently encountered in sound distribution systems are used for illustrating purposes.

Matching at the Amplifier Input

Matching microphones to the input of an amplifier usually means that the impedance of the *source* connected to the input of the amplifier should have the value for which the amplifier was designed. For example, if an amplifier has a rated source impedance of 30 ohms, then the source to connect to the amplifier should have an impedance of 30 ohms. This does not mean that the internal *input* impedance of the amplifier is 30 ohms. In fact, in most typical amplifier designs, the actual internal input impedance of the amplifier will be considerably higher than its rated source impedance. This is nothing to worry about, however; the amplifier manufacturer has said that he wants 30 ohms connected to the input of his amplifier, therefore, the amplifier will operate properly if it is so terminated.

Unfortunately, another factor confuses the issue. A microphone manufacturer may rate his microphone as having a 30-ohm impedance. When actually measured, the microphone may have an impedance of perhaps 40 ohms. Microphones are often rated for impedance in terms of popularly accepted values and not by the actual impedance of the microphone. An impedance mismatch of two to one will not usually cause serious trouble with the frequency response and will make only a slight difference in overall gain; thus the 40-ohm microphone will work with the 30-ohm amplifier.

If there is a transformer within the microphone housing or in the amplifier, serious impedance mismatching may result in loss of either high-frequency or low-frequency response. For example, consider a 300-ohm microphone connected to an amplifier having an input transformer and designed for a 30-ohm source impedance. In this case the low frequencies may be attenuated appreciably, since the transformer was designed to have the correct frequency response when working from 30 ohms. If this amplifier has a high internal input impedance, the output level from a 300-ohm microphone will be 10 db higher than the level of an equivalent 30-ohm microphone, which would be an advantage if the frequency response of the amplifier were not impaired.

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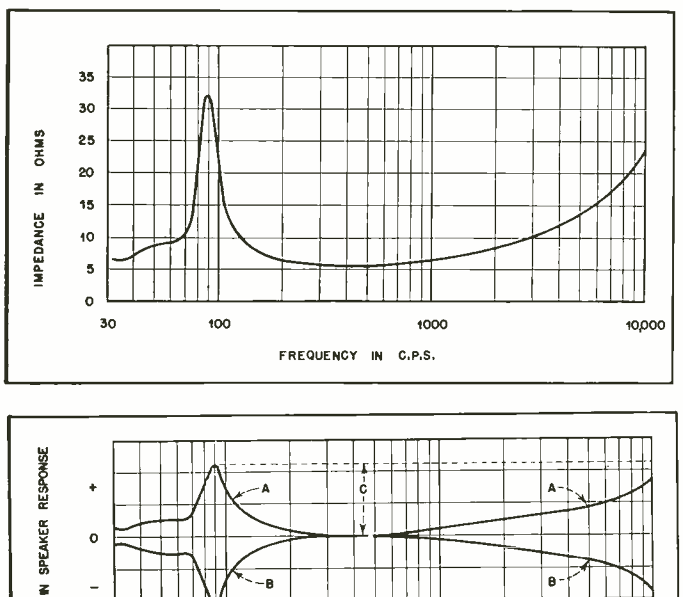


Fig. 1A. (top). Impedance characteristics of a typical cone speaker. Fig. 1B. Effect of amplifier on response of speaker with impedance shown in Fig. 1A. A. Connected to amplifier with output voltage regulation higher than used for original measurement. B. Connected to amplifier with output voltage regulation lower than used for original measurement. C. Maximum change cannot exceed the change in the amplifier output voltage regulation.

Power Gain

When the power gain of an amplifier is computed, it takes into account the effect of the actual input impedance. As stated, most amplifiers have an input impedance higher than the rated source impedance. If the source were working into a matching impedance, the voltage across this matching impedance would be one-half of the open circuit voltage of the source. However, when working into a typical amplifier, the actual input impedance of the amplifier will be high enough so that very nearly full open-circuit voltage will be obtained from the source. These two extremes represent a 6 db change in the available voltage output of the source. Amplifiers are rated for gain by comparing the actual output of the amplifier to the power that the source would deliver to a matching resistance. Since the input voltage to the amplifier may be as much as 6 db higher than the voltage across this matching resistance, as much as 6 db may be included in the gain of the amplifier which is due entirely to the fact that the amplifier's input impedance is higher than its rated source impedance.

The justification for this is that if the amplifier is connected to a microphone and speaker, a certain output will result from a given acoustical input. It does not matter whether the gain which produces this output is entirely tube gain or whether part of it results from having a high input impedance in the amplifier. In fact, by just connecting a microphone and a speaker, it would be impossible to tell whether the gain was tube gain or resulting from the high-impedance input. This explains why a high input impedance is commonly used for amplifiers—as much as 6 db extra gain is obtained.

Matching the Amplifier Output

At the output of the amplifier, the manufacturer indicates the value of load impedance which he desires connected to a given output tap. This impedance is selected to give the maximum power output from the amplifier at the rated total harmonic distortion. This is not the value of impedance to be used in order to obtain the maximum power output. Some other value of impedance will probably give a higher output power when the amplifier is driven beyond rating and distortion is disregarded. This point is frequently confused because it is commonly known that the maximum power is drawn from a circuit when the circuit is loaded with a resistance equal to its own resistance. This is matching for maximum power. However, maximum power is not always the criterion for matching. Most amplifiers are designed for a load impedance approximately two and one-half times the internal output impedance of the amplifier and are designed so that when this load impedance is connected, maximum "undistorted" power will be secured from the amplifier. "Undistorted" in the sound distribution case usually means 5% r.m.s. distortion.

Table I shows what happens, in a typical case, if load resistances of the correct value, twice this value, and half this value are connected to an amplifier. It is apparent from the table that when the amplifier cannot be matched properly, it is better to connect a higher load impedance than a lower load impedance. It also shows that even a two-to-one mismatch does not affect the output as seriously as might be expected.

In many cases, speakers may be switched on and off the output of an amplifier. When all of the speakers are connected in parallel, the load impedance should be equal to or slightly more than the rated load impedance. As the speakers are disconnected, if the amplifier is operated within ratings, there will be no increase in distortion apparent in the remaining speakers, and if the output voltage regulation of the amplifier is low, no apparent change in level occurs.

To determine the power into any speaker, there is a very simple rule: Divide the rated load impedance of the output tap by the rated impedance of that speaker (or the tap on the speaker transformer). This is the fraction of the total power output of the amplifier delivered to that speaker. For example, if a 1,000-ohm speaker is connected to a 500-ohm output tap, that speaker will draw one-half of the power delivered by the amplifier. Thus, by proper selection of loudspeaker impedance, different amounts of power can be delivered to a number of speakers, all connected in parallel to the same amplifier. The rule applies equally well whether the speakers are all connected to the same output tap, or if they are connected to several different-impedance taps. The sum of all these fractions of total amplifier output power should, of course, be equal to one for the best matching. If the impedances of available taps do not permit perfect matching, the sum should preferably be less than one.

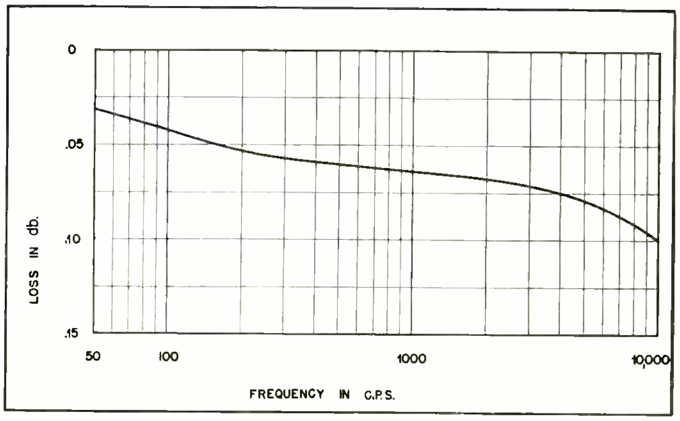


Fig. 2. (right) Curve showing loss per mile. Transmission equivalent for 10 guage copper wire over broadcast range.

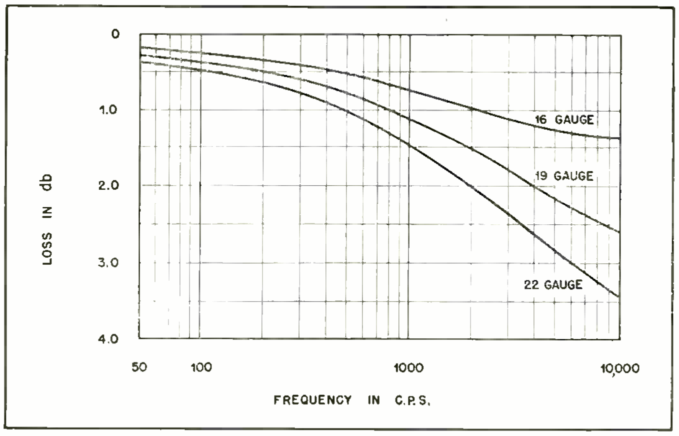


Fig. 3. Curve showing loss per mile non - loaded cable transmission equivalents over broadcast range.

Another method of matching speakers has been coming into favor in recent years and is embodied in current R.M.A. standards proposals. This method uses a standard distribution voltage (the output voltage of the properly loaded amplifier). Speakers (or the taps on the speaker transformers) are then rated not in ohms but in the watts they will draw from this standard distribution voltage. The R.M.A. proposal gives 70.7 volts as one standard distribution voltage. The taps on speaker transformers would be marked ¼, ½, 1, 2, 4, 8—etc. watts. In this system, speaker matching is as easy as connecting light bulbs to a 110-volt power line. The only precaution necessary is to be sure that the total power drawn by all the speakers connected in parallel does not exceed the amplifier rating. The best match in this system is when the total power drawn by the speakers equals the amplifier rating.

Matching Loudspeakers

Table I shows that some mismatch at the amplifier output will not affect the amplifier output too seriously, but the loudspeaker performance must be considered also.

A loudspeaker designer must assume a source impedance to which his speakers will be connected. He knows that the internal output impedance of most amplifiers will be less than the actual impedance of his speakers and designs accordingly.

The actual internal *output* impedance of an amplifier as compared to its rated *load* impedance determines the output-voltage regulation. This may be expressed as the number of decibels the output voltage will increase when the proper load impedance is removed and the output is left on open circuit. This is the maximum

TABLE I

Effect of Load Impedance on Amplifier Output and Distortion

Load Resistance Connected to 500-ohm Output	Total Harmonic Distortion at 400 cps	Output Power
(A) (Input voltage and control settings held constant)		
500 ohms	5% r.m.s.	32.2 watts
1,000 ohms	8% r.m.s.	25.0 watts
250 ohms	13% r.m.s.	24.3 watts
(B) (Input voltage varied to secure maximum output at 5% distortion)		
500 ohms	5% r.m.s.	32.2 watts
1,000 ohms	5% r.m.s.	22.5 watts
250 ohms	5% r.m.s.	16.4 watts

(Data taken on production sample 25-watt Stromberg-Carlson AU-33 Amplifier)

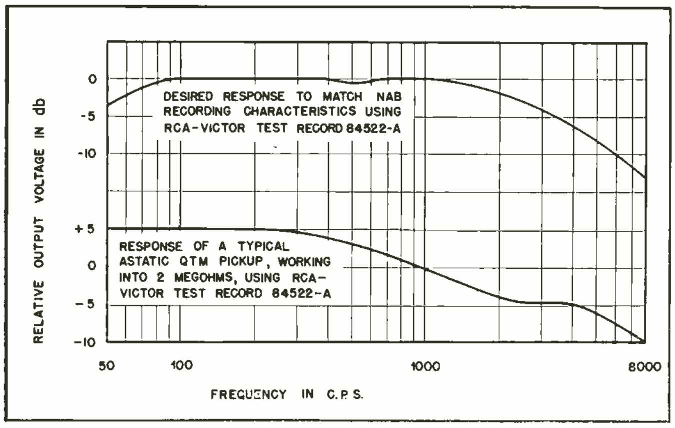


Fig. 7. Comparison of pickup response and NAB recording characteristic curve.

change in level to be expected as speakers are switched on and off the amplifier output. The equation is:

Output-voltage regulation in db =

$$20 \log_{10}\left(1 + \frac{\text{Output Impedance}}{\text{Load Impedance}}\right)$$

For an output-voltage regulation of 3 db, this equation gives a ratio of output impedance to load impedance of two-fifths. Six db means that the output impedance and the load impedance are equal. Thus the lower the output-voltage-regulation figure, the lower the output impedance.

The impedance of cone loudspeakers is a function of frequency as shown in *Fig. 1*. The rated value of speaker impedance is usually about equal to the actual impedance in the 400-1000 cps region. Thus, over much of the frequency range, the impedance of the speaker is considerably higher than the rated impedance.

In the past it has been a common practice to measure the frequency response of loudspeakers from a zero-impedance source (i.e. zero voltage regulation). The speaker may be measured using any value of source impedance specified by the speaker manufacturer. In any case, it is obvious that if the output impedance of the amplifier differs greatly from the source impedance used in measuring the speaker response, the speaker will have a frequency response when connected to the amplifier different from the response obtained by measurement on the speaker alone. Since speakers for sound distribution systems are designed to work from a low impedance—usually less than the nominal impedance of the speaker—an amplifier to drive a speaker must have a low output-voltage regulation (*i.e.* low output impedance) to give the expected performance. If the output-voltage regulation of the amplifier is high, the bass and treble response of the speaker will be exaggerated—but never by more than the output-voltage regulation of the amplifier in decibels, as shown in *Fig. 1b*.

Likewise, it is not advisable to connect many speakers in series. At each frequency, the impedance of the individual speakers will be slightly different, and thus the voltage drop across each speaker will vary. This will put more variations into the frequency response. Occasionally, however, there may be some reason for connecting 3 or 4 speakers in series, and if they are all alike, the results will be satisfactory.

One of the requirements to secure good results at the speaker is the proper choice of a speaker transformer. To have a good frequency response, the transformer must have a good frequency response. To have low distortion, the transformer should not create distortion. Typical sound distribution transformers have a frequency response that is flat within 3 db between 70 and 10,000 cps. The power-handling capacity of transformers, however, decreases as the frequency is reduced. Therefore, two (or more) ratings are generally provided for these transformers. For example, the Stromberg-Carlson RT-30 can carry 5 watts above 70 cps or 10 watts above 100 cps. The RT-40 can handle 15 watts above 70 cps or 25 watts above 100 cps. These transformers have high efficiency—the insertion loss for each transformer is only 1 db.

Speaker transformers should be designed with a high primary inductance. If a large number of transformers with low primary inductance are connected in parallel, the net effect is a low-impedance load on the amplifier at low frequencies. This may overload the amplifier at low frequencies, producing distortion; and of course it will always reduce the low-frequency response, not just for the speakers using the poor transformers but for the whole speaker distribution system. The use of a speaker transformer with a poor low-frequency response is *not* a good way to provide attenuation of the frequencies below the cut-off of a horn speaker. The open circuit primary reactance at the lowest frequency of transmission should be at least 2 or 3 times the rated primary impedance with the secondary properly loaded.

Matching Lines

Transmission lines may be considered in three groups. The very short line may be considered as only a pair of wires. The very long line must be considered with all of the care that A. T. and T. bestows upon its long-distance trunks, and in between are moderate-length lines which require an intermediate engineering approach.

In the short line, the only problem is the loss due to the resistance of the line. Table II gives the maximum length of various wire sizes and load resistances to keep the maximum line loss less than one db. To convert this table to ½ db loss, divide the length of line by 2. Multiplying line length by 2 converts the table to 2 db loss. For very short lines, the loss is independent of frequency. Another useful guide is given in Table III.

The very long transmission line has a "characteristic impedance." This is the impedance measured across the two terminals of a very long length of the line. It is necessary to terminate the line at each end in an impedance equal to this characteristic impedance. Failure to do so will result in echoes or reflections from the terminations which will garble the program being transmitted. Further,

[*Continued on page 37*]

TABLE II
Wire Gauge Data

Wire Gauge	Amplifier Output Taps: 4 ohms	8 ohms	16 ohms	250 ohms	500 ohms
22	15 Ft.	30 Ft.	60 Ft.	1000 Ft.	2000 Ft.
20	25 Ft.	50 Ft.	100 Ft.	1600 Ft.	3200 Ft.
19	30 Ft.	60 Ft.	120 Ft.	2000 Ft.	3700 Ft.
18	40 Ft.	75 Ft.	150 Ft.	2500 Ft.	5000 Ft.
16	60 Ft.	120 Ft.	240 Ft.	3700 Ft.	6200 Ft
14	100 Ft.	200 Ft.	400 Ft.	6200 Ft.	10000 Ft.

Values Rounded

TABLE III

Ratio of Load Impedance to Line Resistance	*Maximum Line Loss
1	6 db
2.5	3
4	2
8	1
16	0.5

*The maximum loss occurs when the source is zero impedance.

MUSICAL ACOUSTICS

BENJAMIN F. TILLSON*

PART VII

This is the seventh of a series of articles on music theory, written especially for sound engineers.

IN PART VI attention has already been directed to the variation in width of cut grooves because of the angular relationship of the direction of stylus travel to the plane face of its cutting edge.

Unless it were practical to maintain a constant orientation of the cutter to the direction of its track at any instant, or unless revolving milling cutters were practical, this appears to be a fundamental disadvantage of all cut grooves, because the two bounding walls must then be offset, not parallel, curves.

Except when such walls are distorted by pressure, no reproducing stylus of circular cross-section can make simultaneous contact with both walls unless it travels in an oscillatory manner normal to the disc surface and has a configuration which will offer differently dimensioned sections to the groove.

Such oscillations must occur twice in every fundamental wavelength, so they encourage second harmonics and higher even-numbered harmonics where shock or inertia are associated with them.

Fig. 5 shows an exaggerated form of offset inflected curves, and the maximum diameters of circles which can pass between them. The offset of the two outside curves is twice that of each component channel, but the larger channel will pass a circle more than twice the diameter of those which will pass through its component channels. There is also a difference in the angles which the contact diameters make to path of travel.

The statement commonly appears that commercial records have abrasive fillers to grind the stylus to a shape which will fit the groove. Such grinding may permit a large tip radius of a needle to bottom in a groove eventually. But how can it be made to fit between non-parallel walls at all points? Such can be done only by destroying the original cut form of those walls with an associated loss of the message they bore; or else by a pressure distortion of those walls which may not then center the reproducing stylus as was centered the cutting stylus. The abrasion of the tip will destroy its spherical form and circular cross-section which, by displacement vertical to the disc surface, might otherwise permit bearing against both walls of the groove.

Consequently, lateral-cut records probably have a stylus bearing against only one wall of the groove at any instant. If the outer wall guides the reproducing stylus on its rise up a sound wave, the inner wall will direct its drop. At each crest and valley there must be a changeover from outer to inner groove walls, and vice versa. Such must be points of vibrational shock which will also occur twice every wavelength. This makes four uniformly spaced harmonic disturbances per wavelength; and they are probably of a nature that would originate higher harmonics and unharmonic "clang" tones. The nature of these switching paths is illustrated in *Fig. 6*. The arrows show direction of travel of recording and reproducing stylus.

In that figure the original sound curve of the center line of the lateral-cut stylus is marked *A*. One of the stylus edges cuts the offset wall-curve *B*, which a circular-section reproducing stylus must follow. And the curve *C*, traced by the center line of the reproducing stylus, represents what can be reproduced from the record. Curve *C* can be seen to diverge from the original curve *A*. The former's radii of curvature are much shorter in the valleys, and longer above the crests; so the two half cycles are no longer the same in uniformity, and that means a distorted modulation.

Just as will be shown for vertical-cut record tracking in Part VIII, there is a phase displacement of amplitudes because the point of contact (tangency) of the reproducing stylus is not in the line of its axis, or of that of the cutting stylus originally where the bearing is against only one wall of the groove.

Value of Coordinates

The equation of the reproduced curve is entirely too complicated to express here, but the value of its coordinates can be given in terms of a prototype sine curve: $y = a \sin bx$.

The cutting stylus merely lowers that curve a distance down the Y-axis equal to one-half the cutter width, keeping the new X-axis parallel to the original X-axis. The lower wall of the groove is then represented by the sine curve $y_1 = a \sin bx_1$. The tangent to the sine curve at any contact point x_1 equals the first derivative $dy_1 = ab \cos bx_1$.

The circular cross-section of the playback stylus has a radius r and its center traces the path of the reproduced tone. Its radius normal to the tangent makes an angle θ to the Y-axis, which is equal to the angle θ which the tangent makes to the X-axis.

Calling the coordinate of the center x_2, y_2 when the circle is tangent to the groove at point x_1, y_1, the following relationships hold:

$x_2 = x_1 \mp r \sin\theta$ and $y_2 = y_1 + r \cos\theta$. See *Fig. 7*.

In these equations $b = 2\pi/\lambda$ and $a =$ amplitude.

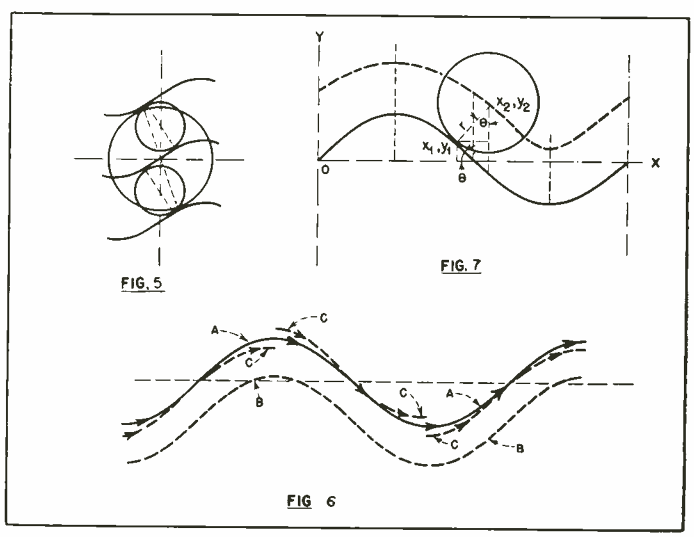


Fig. 5. Maximum diameter of circles within offset inflected curves. Fig. 6. Groove switching paths. Fig. 7. See text.

*Consulting Engineer, Montclair, N. J.

From trigonometry we know that

$$\sin \Theta = \frac{\tan \Theta}{\sqrt{1 + \tan^2 \Theta}}$$

$$\text{and } \cos \Theta = \frac{1}{\sqrt{1 + \tan^2 \Theta}}$$

$$\text{Therefore, } \sin \Theta = \frac{ab \cos bx_1}{\sqrt{1 + a^2 b^2 \cos^2 bx_1}}$$

$$\text{and } \cos \Theta = \frac{1}{\sqrt{1 + a^2 b^2 \cos^2 bx_1}}$$

By inspection of the figure it will be seen that the center of the circle is at all times above the track on which it rests, so that always $y_2 = y_1 + r \cos \Theta$. But the center of the circle is sometimes to the left of its tangent point, and sometimes it is to the right. In the former case $x_2 = x_1 - r \sin \Theta$, and in the latter case $x_2 = x_1 + r \sin \Theta$.

Bearing these conditions in mind we can determine the coordinates of the trace of the center of the reproducing stylus as follows:

$$x_2 = x_1 \mp \frac{r\, ab \cos bx_1}{\sqrt{1 + a^2 b^2 \cos^2 bx_1}}$$

$$y_2 = a \sin bx_1 + \frac{r}{\sqrt{1 + a^2 cb^2 \cos^2 bx_1}}$$

The falsity of the reproduced path also causes a variation in the velocity of the reproducing stylus, along that path, which is not in accord with the velocities of the cutting stylus.

The cutting stylus has V-shaped walls usually forming an included angle of 90° to 87°, but only at the crest and at the valley, or at some nearby points if it is pitched at a clearance angle not normal to the unmodulated sound track, will the sound groove it cuts have the same angle of trough. The face of the cutting stylus varies in its orientation to the path of

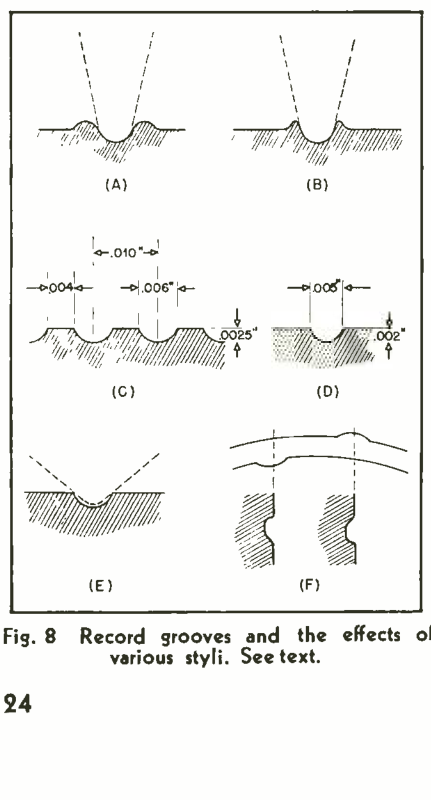


Fig. 8 Record grooves and the effects of various styli. See text.

the sound track; and if we call such angle Ø, also the angle of the cutting stylus α, and the resultant angle of the groove β, then the following relationship holds: $\tan \frac{\beta}{2} = \sin Ø \tan \frac{\alpha}{2}$ and its effect is shown in the tabulation:

Value of α	Values of β for Ø = 30°	45°	60°
90°	53.2°	70.5°	81.8°
87°	50.8°	67.8°	78.8°

Since the value of Ø is continuously and periodically varying in a sine curve, the cutter is scooping out undulations in the side walls which undulations were not present in the original wave and must cause a series of harmonics to be added to the fundamental. If the cutting stylus had a non-radial feed, this situation is further complicated by varying eccentricities of the cutter-head arm. But such a difficulty does not seem to appear in the making of vertical-cut records.

When one realizes the complexity of the recorded sound track and its more abrupt changes than those of the simple sine curve, it is obvious that the distortion is even greater but may not be identified because of the masking of tones in the resultant confusion. Besides, there is also the noise from the reproducer stylus as it gouges into the record at the pinches.

Then, too, the unbalanced, one wall, bearing pressures in the groove cause at the least a resilient compression of the record surface material; and the elastic response to such varying pressures may produce adventitious harmonic oscillations. Or, if the elastic limit of the record material be exceeded, there is a destruction of the surface and its fissuring with cracks; or else a permanent change in the true recorded form of the sound groove either by distortion or wearing off of the finer variations of the higher harmonic frequencies. Such jamming of the pickup stylus in the groove is necessary to prevent its uncontrolled rattling around with consequent inharmonic frequencies typical of noises.

Identical Recording-Reproducing Feeds

An English book of 1933 describes various home recorders and states that most gramophone turntables are "dished" downwards, therefore a carrier disc is carried upon the turntable to provide a firm foundation for the superimposed record blank.

Various lead screw drives feed the recording head inwards or else such feed was controlled by a spiral groove in the carrier disc, which then became a tracking disc as exemplified by the Kingston-Wearite Recorder. In it the tracks are an outer annulus and the recording blank is placed centrally within them. Or in the S. A. Jennings Acoustic Recorder of 1916 the tracking disc occupied the inner central position and the outer annulus was for sound recording at a greater linear speed. Each had an auxiliary tracking arm attached to the recorder-reproducer arm; and the tracking arm carried a thorn needle stylus spring-mounted so that the necessary weight on cutting needle was not affected. To rear of the recording arm was an adjustable weight so that during play-back the pressure on the stylus could be relieved.

The tracking point was gradually forced inwards, due to the spiral on the tracking disc, and carried with it the whole of the cutting (embossing) arm. These designs seem to be an anticipation of the feed described in the May 1947 issue of *Audio Engineering* for the Wagner recorder, except that the latter's tracking grooves were on the reverse side of the sound record blank or else on a carrier disc. That design has since been abandoned in favor of a leadscrew feed for recording and reproducing.

Pre-grooved Records

The same 1933 English book speaks of a recent revival in America of pre-grooved records for home recording because of a special synthetic resin pre-grooved record

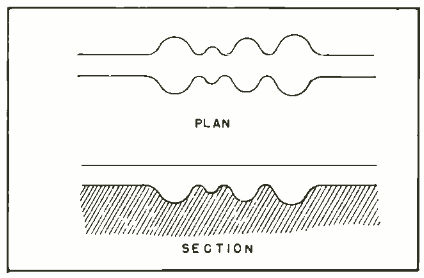


Fig. 9 Embossed track for vertical recording.

introduced by the R.C.A. Victor Co. It claims that celluloid was too soft and wearable, and that pre-grooved aluminum or other metal discs gave greater surface noise or scratch.

With plain-surface aluminum discs the recording stylus was a plain steel needle, more obtuse at the tip than an ordinary gramophone needle, which gives a scribing action, pushing away the aluminum surface to leave it with humped ridges on each side of the groove, as in *Fig. 8a*. Or if a diamond stylus were used leaving sharp burrs on the sides of the groove as shown by *Fig. 8b*. The playback needle rested on the bottom of the groove, and is indicated by the dotted lines.

The Standard record grooves at that time are shown by *Fig. 8c;* the R.C.A. pre-grooves were as in *Fig. 8b;* and the needle employed for recording had a very obtuse point which exerted considerable pressure on the edges of the groove. Because of the rotation of the disc these edges are embossed or pressed down somewhat as shown in *Fig. 8e*. Thus the recording takes place on the upper edges of the groove, not at the bottom which remains a true spiral, producing an effect as in *Fig. 8f*.

The same recording needle was used for playback. Although the results were thought to be not so perfect theoretically as were those where cutting or scribing was effected straight into a plain disc, yet those Victor discs were credited with considerably better results than those of plain aluminum. This was in the era when fibre needles were extensively used and probably offered a low standard for comparison.

It is quite obvious from *Fig. 8f* that neither the use of the recording stylus nor any other type of stylus for playback assures a fit of it to the sound groove where lateral recording is employed. The stylus would bounce back and forth from one side wall to the other, thus contributing to distortion.

This situation would not prevail with vertical recording, where an embossed track might resemble *Fig. 9*. But one might well question whether vertical embossing would be of sufficient magnitude to reproduce any great range of frequencies and intensities.

A vertical-cut track in a pre-groove would have the same errors as will be discussed in the following article, and similar to those discussed for lateral-cuts earlier in this article.

Radius of Curvature of Sound Undulations

Because of the relationship of stylus diameter to its amplitude of swing and to the sound track velocity, it is of interest to determine the minimum radius of curvature for any sine curve which represents a pure tone as in *Fig. 10 (a)*, and to offer mathematical support of the conclusion that the ratio of such radius of curvature of the center line of a sine sound track to its maximum amplitude from neutral equals the square of the quotient of the wavelength divided by two pi and such ratio equals 0.02533 times the square of the wavelength. (λ).

In differential calculus we have learned that the first derivative of the equation of a curve expresses the slope of tangents to such a curve with respect to the *X*-axis, and that when such slope equals zero the tangent is parallel to the *X*-axis at points which represent maximum or minimum ordinates to the curve; and that when the second derivative has a value less than zero that ordinate is a maximum or crest in the curve.

In *Fig. 10a* the relative movement of the coordinate origin O to the left a distance $\frac{c}{b}$ simplifies the equation of the curve to $y = a \sin bx$ where $b = \frac{2\pi}{\lambda}$; $\tan \emptyset = \frac{dy}{dx} = ab \cos bx$; for $\tan \emptyset = 0$; $\cos bx = 0$ and bx must equal odd numbers of $\frac{\pi}{2}$, or 90°, 270°, etc. When $bx = \frac{\pi}{2}$; $\frac{2\pi x}{\lambda} = \frac{\pi}{2}$; so $x\ \frac{\lambda}{4}$ and $\frac{dy}{dx} = ab \cos \frac{b\lambda}{4}$ The second derivative $\frac{d}{dx}\left(\frac{dy}{dx}\right) = -ab^2 \sin bx = -ab^2 \sin \frac{b\lambda}{4}$ where $x = \frac{\lambda}{4}$. This equals less than zero so the point is a maximum. The customary expression for any radius of curvature ρ is

$$\rho = \left[1 + \left(\frac{dy}{dx}\right)^2\right]^{3/2} \div \frac{d}{dx}\left(\frac{dy}{dx}\right).$$

Thus by substitution $\rho = [1 + a^2 b^2 \cos^2 bx]^{3/2} \div - ab^2 \sin bx =$

$$\frac{\left[1 + a^2 b^2 \cos^2 \frac{b\lambda}{4}\right]^{3/2}}{- ab^2 \sin \frac{b\lambda}{4}}$$

but $b = \frac{2\pi}{\lambda}$; so

$$\rho = \frac{\left[1 + \frac{4\pi^2 a^2}{\lambda^2} \cos \frac{\pi}{2}\right]^{3/2}}{- \frac{4\pi^2 a}{\lambda^2} \sin \frac{\pi}{2}}$$

and since $\sin \frac{\pi}{2} = 1$ and $\cos \frac{\pi}{2} = 0$, therefore

$$\rho = \frac{1}{\frac{4\pi^2 a}{\lambda^2}} = - \frac{\lambda^2}{4\pi^2 a} = - 0.02533 \frac{\lambda^2}{a}$$

The negative sign shows that the curve is convex upwards at this point of maximum.

It would appear that there is virtue in making embossing styli which have much smaller tip radii of as little as ¼ mil. They have been successfully used with radii of ½ mil. One would not expect to find details in a picture made with finger or palette knife used to apply the colors and yet the over-all effect may have a certain color charm, when viewed from a distance. Musical recordings may be in the same situation because of the grossness of the styli used to make and play back records. Any advance in high fidelity will require that attention be given to that condition.

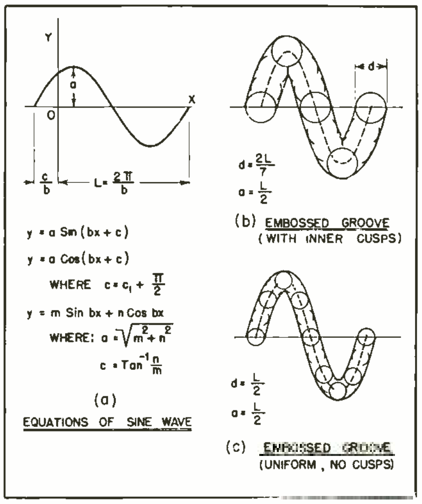


Fig. 10. In (a), method of determining minimum radius for sine curve, (b and c), embossed grooves with and without inner cusps.

Fig. 10b shows how a sudden widening may occur at the crests of the waves of an embossed sound track with angular cusps on the track's inner wall where the width of the track is too great for the pitch of its modulations. This will promote distortion from the uncertainty of the reproduced maximum amplitudes of any swing duplicating those recorded. And *Fig. 10c* shows how such a condition is avoided when the sound track is narrower.

Where the radius of the stylus is too large for the minimum radius of undulation curvature of the groove, as it often may be, it will be necessary to make some assumptions in order to calculate the new relationship to that of a proper radius of stylus cone.

Slopes of Sine Curves

Let us assume that it will bear against the outside of the groove eight-ninths (almost 90%) of the time, and let us also assume three different conditions for the maximum slopes in the undulations of the sound groove axis; namely, 30°, 45°, and 60°.

Then $bx_1 = \frac{4\pi}{9} = 80°$ and $x_1 = \frac{2\lambda}{9}$ therefore the radius of curvature $\rho = [1 + 1.19042 \frac{a^2}{\lambda^2}]^{3/2} \div - 38.87 g \frac{a^2}{\lambda^2}$

Table III shows ratios corresponding to certain maximum slopes of sine curves.

The narrower sound track promotes higher fidelity at a lower linear speed and a greater playing time on a given diameter of disc with the same amplitude of stylus swings; and it also permits a greater number of sound tracks per inch radius of disc because so great an amplitude of stylus swing is not needed for equal high fidelity of sound reproduction. As many as 515 sound tracks per radial inch (1.94 mils center line spacing) have been successfully embossed on a disc of 1½ inch minimum diameter; and played at 33-⅓ r.p.m. it has an unmodulated track speed of 2.618 inches per second. It is said to have recorded and reproduced a frequency of 11,000 cycles per second. This wavelength on the above sound track would be only 0.238 mils and the ampli-

Table III

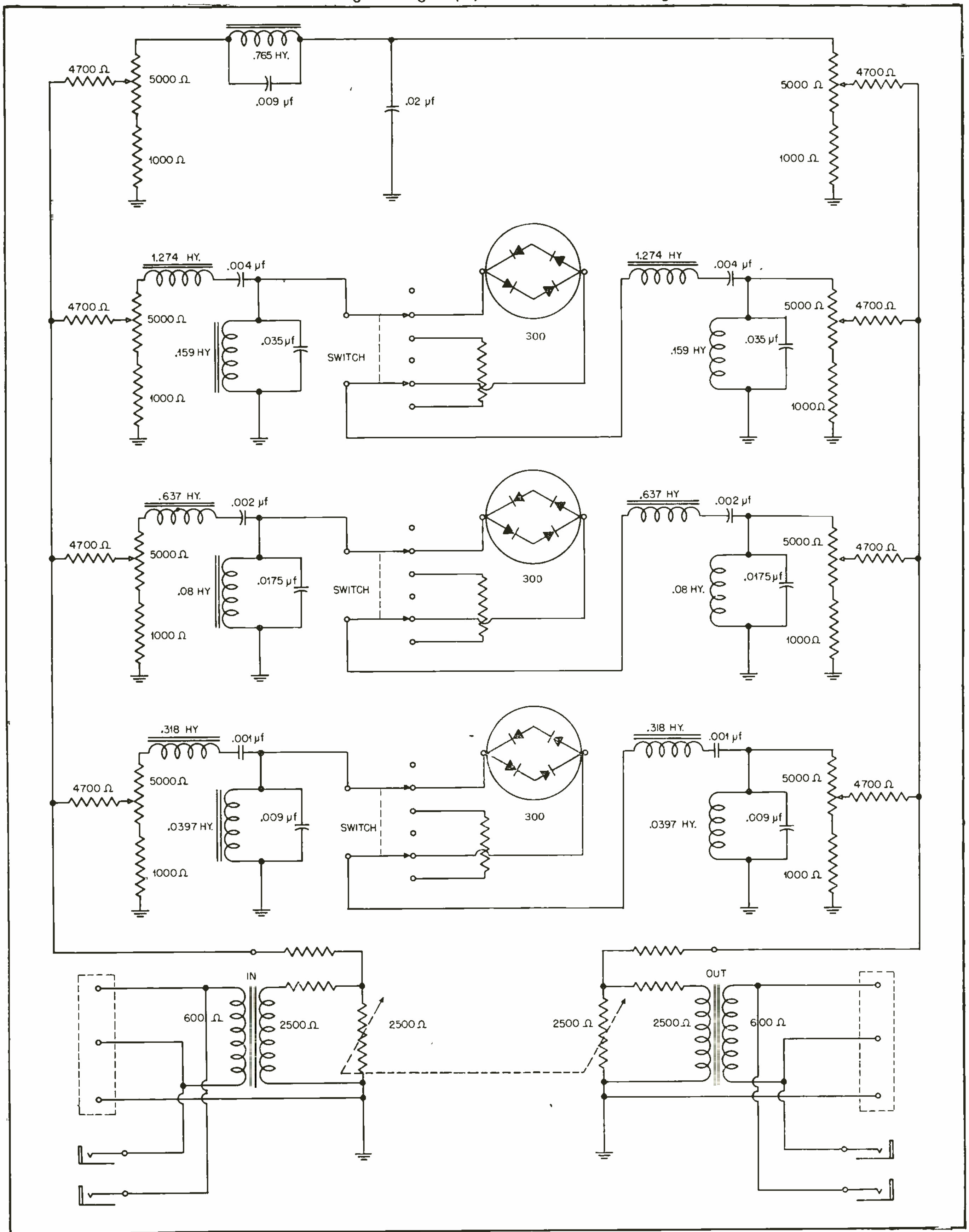
Max. Slope of Sine Curve	Ratio $\frac{a}{\lambda}$	$\frac{\lambda^2}{a}$ at point x equals		The Ratio of Radius of Curvature to a at point x equals λ at point x equals	
		λ/4	2λ/9	λ/4	2λ/9
30°	0.1000	0.02533	0.026182	0.002533	0.002618
45°	0.1592	0.02533	0.026894	0.004033	0.004282
60°	0.2757	0.02533	0.029290	0.006984	0.008075

Table IV

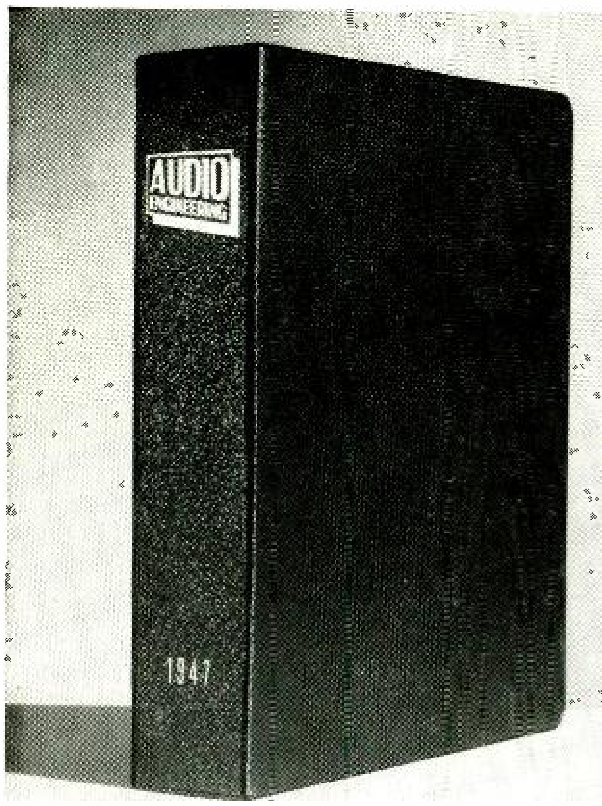
Max. Slope	Ratio a/λ	Max. Slope	Ratio a/λ
10°	0.0281	50°	0.1897
20°	0.0579	60°	0.2757
30°	0.0919	70°	0.4372
40°	0.1339	76°	0.6383
45°	0.1592	80°	0'9014

Schematic Diagram of Experimental Noise Suppressor

CHESTER D. COLE
General Engineering Dept., American Broadcasting Co.



This noise suppressor was developed by the General Engineering Dept., American Broadcasting Co., and was described at the recent NAB Convention in Atlantic City. A complete article on this suppressor will appear in Audio Engineering next month.



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NEW PRODUCTS

SINE WAVE CLIPPER

A new Sine Wave Clipper has just been announced by Barker & Williamson, Upper Darby, Pa., which will be considered an important addition to the laboratory equipment of most engineers interested in audio frequency circuits.

This newest member of the popular line of B & W specialized test instruments provides a test signal particularly useful in examining the frequency response and transients of audio circuits. Designed to be driven by an audio oscillator the B & W Clipper provides a clipped sine wave—hence the name "Sine Wave Clipper."

By feeding the output of the clipper into audio equipment under test and in turn introducing the equipment's output into an oscilloscope, the experimenter or engineer may quickly view and analyze distortion introduced by the amplifier.

Barker & Williamson engineers have photographed and classified the typical kinds of distortion viewed on the oscilloscope when testing equipment in conjunction with the clipper. These photographs are reproduced in the instruction book accompanying each Sine Wave Clipper.

This new device is a great time saver when used in engineering work, repairs or on equipment under development. A sine wave analysis after every change in a component becomes time consuming and tedious. By means of the Sine Wave Clipper, however, the effect of making changes in a circuit may be seen instantly and thus guide the course of development in the proper direction.

The routine use of the clipped sine wave, in addition to sine wave measurements, makes for a more complete check on the stability of equipment in regular operation.

Complete information on this new device is available from the manufacturer.

RCA PM SPEAKERS

A standard line of PM speakers for general replacement and sound-systems work has been announced by the Renewal Sales Section of the RCA Tube Department.

The line includes a "controlled-resonance" 12-inch speaker, a 4-inch and a 5-inch speaker, a 4 by 6 inch elliptical speaker, and a 2 by 3 inch elliptical speaker.

Rated at 12 watts power-handling capacity, the 12-inch speaker has a one-piece stamped steel frame for rattle-free operation, and a unique filter which automatically filters needle scratch and other objectionable high-frequency noises.

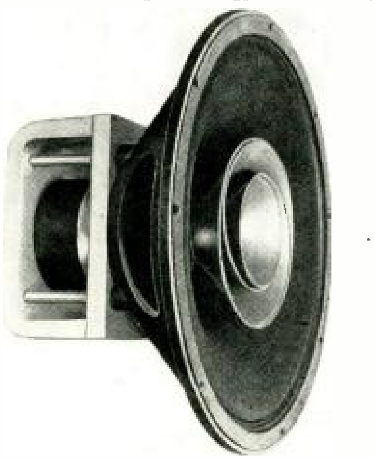
The RCA 4 and 5-inch speakers in the line are specially designed to meet the radio serviceman's small radio requirements.

The RCA 4 by 6 inch and 2 by 3 inch elliptical speakers are designed for table-model combinations and portable radios.

Complete specifications and descriptions of the RCA speakers are included in Catalog sheet 2F384R available from RCA tube and parts distributors, or the Renewal Sales Section, Tube Department, Harrison, N. J.

STEPHENS SPEAKER

In line with the demand for a reproducer to convert existing equipment for high fidelity, the Stephens Manufacturing Corporation offers the newly designed Tru-Sonic Model P-52FR Co-Spiral Speaker embodying novel constructional features. The Differential Diffuser, patent pending, accomplishes high frequency dispersion with an almost 100% spherical polar pattern of over 90°.



Frequency range is from 40 to 14,000 cycles. As a result of numerous tests on representative groups of individuals, frequency characteristic has been calculated to attenuate record motor rumble below 70 cycles, and to emphasize a band in the "power" range around 500 cycles. "Presence" is accentuated by a rise at 2300 cycles. High-frequency "hash" is subdued by a gradual roll-off from 8000 cycles. Supplied in both 12" and 15" cone diameters. Equipped with 2½ lbs. of Alnico V magnet material. Efficiency over 50% available with either 8 or 16 ohm voice coils. 15 watts of power input handling capacity. Available for immediate delivery.

Bulletin No. 109 describing the complete Stephens Tru-Sonic line may be had by writing Stephens Manufacturing Corporation, 10416 National Blvd., Los Angeles 34, California.

KLIPSCH SPEAKER

A new 2-way sound reproducer is being manufactured by Broeiner Electronics Laboratory of New York City. A basically new design, the unique feature of the Klipsch Speaker System is its use of a horn for the low frequencies as well as for the high frequency range. Conventional 2-way speaker systems use a direct radiator for the low frequencies.

The Klipsch low-frequency horn is folded in an ingenious manner, and uses the corner of the room as an integral part of the acoustic system so that, occupying only 14 cubic feet of space, it provides performance equivalent to conventionally designed horns 8 to 16 times as bulky. Such large horns are used in theatre installations, but could hardly be used in a living room, while the Klipsch Speaker System fits unobtrusively into a corner, where it utilizes the converging walls and floor as an extension of the low-frequency horn. The listener is literally inside the loudspeaker. With both the high and low-frequency speakers coupled to horns, their relative efficiencies are nearly equal. Thus, it is not necessary to attenuate the high frequency unit markedly to match the low-frequency output, as in the case of direct radiators.

The gain in efficiency permits the use of a relatively simple, high-quality triode amplifier of moderate cost, for most applications. Because of the horn loading, voice-coil excursion at low frequencies, even at full power, is extremely small. As a result, distortion is of the order of 1/100th to 1/400th that of direct radiators.

The non-resonant character of the speaker system affords reproduction of transients without hangover; there is no resonant frequency in the bass range to be excited by all notes in its vicinity. Bass instruments are clearly recognized. The original tones are *reproduced*; one does not hear sounds *originated* by the speaker.

A novel design of high-frequency horn provides a 90-degree horizontal distribution pattern of frequencies above 500 cycles, to match the dispersion of the low-frequency horn. Exceptionally uniform high-frequency dispersion is accomplished without the usual multicellular construction, which produces a ragged distribution pattern.

Fundamentals from 30 to 15,000 cycles are cleanly reproduced and uniformly distributed throughout the room.

The dividing network is a constant-resistance, parallel type, providing 12 decibels per octave attenuation, and has a cross-over frequency of 500 cycles—sufficiently low to reduce to a negligible value cross-modulation in the low-frequency speaker. An L-pad permits adjustment of balance to suit individual conditions. Air-core inductors are used, to avoid distortion and cross-modulation products ordinarily resulting from non-linear magnetic components.

The Klipsch Speaker System is available as a complete unit, and the individual components such as the high- and low-frequency horns, drivers, and dividing network, can be obtained separately. The Model 1A is rated at 20 watts. Other models provide power handling capability up to 60 watts in one unit.

PRINTED CIRCUIT TECHNIQUES

In response to an unprecedented demand for technical information on printing electronic circuits the National Bureau of Standards has just published the first comprehensive treatment of this subject entitled *Printed Circuit Techniques*, by Drs. Cledo Brunetti and R. W. Curtis. The booklet consists of 10 chapters totalling 43 large, two-column pages and is adequately illustrated with 21 half-tones, 18 line cuts and five tables. NBS Circular 468, *Printed Circuit Techniques*, is now available from the Superintendent of Documents, U. S. Government Printing Office, Washington 25, D. C., at 25 cents per copy.

ALLIED RADIO CORPORATION

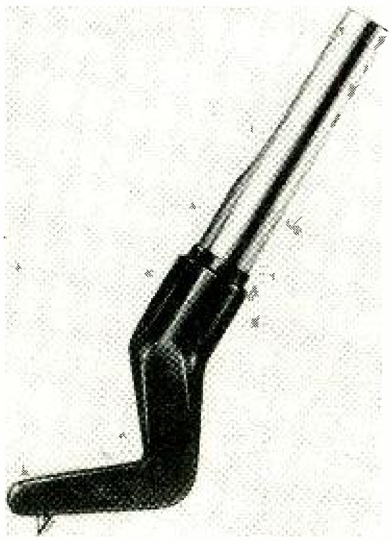
Allied Radio Corporation announces the publication of a new 48-page supplement to their regular master catalog.

The supplement features the most recent developments in radio and electronic equipment as well as latest price information. Included are new wire and disc recorders, test instruments, an added new line of transformers, a television kit and other new builders' kits, high-fidelity sound equipment, new amateur transmitting equipment, communications receivers and radio receiving sets. Latest price information is provided for standard tubes, test units, condensers and resistors, including many price reductions on this and other merchandise regularly used in the radio and electronics field.

The new supplement No. 114 as well as Allied's regular 164-page master catalog No. 112 can be obtained without charge from the Allied Radio Corporation, 833 West Jackson Blvd., Chicago 7, Ill.

PHONO NEEDLE

A rare osmium alloy tip, especially developed for fine tonal reproduction, is used for the first time in the new black nylon phonograph needle now being introduced to the trade by Webster-Chicago.

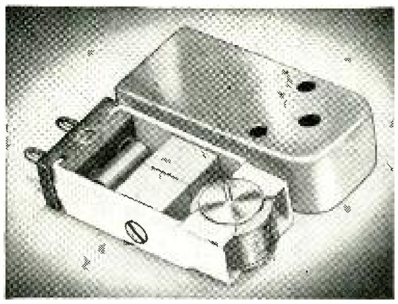


The new metal tip is an alloy, formed by powdered metallurgy on a base of osmium, a rare and expensive mineral element. It is micro-ground to a mirror-like finish which provides a smooth, gentle ride of the needle in the record groove.

The black nylon needle employs the same unique nylon "knee-action" which features the popular Webster-Chicago Ivory Nylon Needle with the Jewel Tip which was introduced so successfully a year ago.

WIDE RANGE MAGNETIC PICK-UP

Offering a new approach to magnetic reproducers, the Clarkstan RV Pickup presents a high-fidelity, wide-range device of extreme simplicity and ruggedness. The stylus can be instantly removed and replaced by the fingers without the use of any tools, thus permitting the use of styli with various ball-point radii. This pickup has a flat frequency response beyond f-m requirements. The needle, which weighs 31 mg, is the armature and is the only



moving part. All needless mass has been eliminated, thereby reducing needle talk to a minimum. The pickup is stable under all normal working conditions of temperature and humidity. High impedance is the standard winding, but it can be had in impedances of 5, 50, 250, and 500 ohms. When used in conjunction with Clarkstan Sweep Frequency Transcriptions, because of its flat response, the pickup will serve as a secondary sweep frequency standard. The pickup is made to sell in conjunction with a transcription-type tone arm. The specifications on the pickup are:

Response. Exactly velocity responsive to 15,000 cps.

Stylus. Sapphire with standard .003" radius ball point, 50° cone angle. Radii of .0015", .0022", .0025" also available.

Needle Force. 20 grams optimum for commercial pressings.

[Continued on page 39]

TECHNICANA

FREMODYNE CIRCUIT

● The Hazeltine FreModyne circuit combines superheterodyne and superregenerative principles to form a sensitive, simple and practical f-m detector. It is the result of extended theoretical and practical development work, and is intended primarily for addition to low-price a-m receivers in order to bring f-m programs within the reach of all income groups.

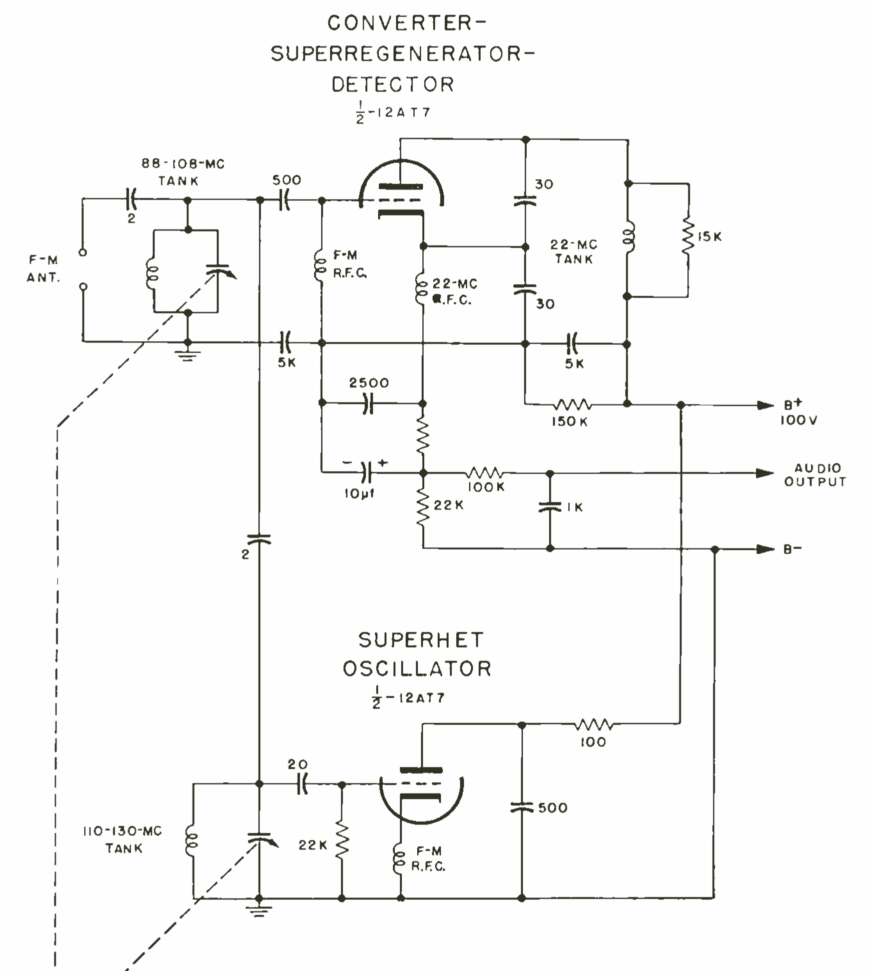
The circuit utilizes only one dual triode to convert the relatively weak f-m signals from the f-m antenna into an audio signal voltage which is large enough to operate the conventional audio system of a-m receivers.

In the FreModyne circuit one triode of the dual-triode tube serves merely as the local oscillator necessary for superheterodyne frequency conversion. The other triode performs four functions, operating as (1) a superheterodyne converter to an intermediate frequency of about 22 megacycles, (2) a superregenerative i-f amplifier of high gain, (3) a converter from f.m. to a.m. and (4) a detector delivering audio output. For brevity this triode is hereafter referred to as the superregenerator. The f-m signal is converted to a.m. by side-tuning the receiver.

The use of the superheterodyne principle in the FreModyne circuit greatly reduces signal-frequency radiation compared to a conventional superregenerator (approximately 30-40 decibels reduction), and provides more uniform superregenerative operation. The circuit also includes a special automatic stabilizing arrangement permitting the regeneration control of the normal superregenerative receiver to be discarded. This stabilizing circuit also permits a quench wave of special shape to be obtained which gives good selectivity, good audio output, and quite linear f-m detection.

Being side-tuned for f-m reception, a FreModyne receiver has two responses for each station, both of which represent correct tuning. This compares with three responses in conventional f-m receivers, only one of which represents correct tuning. The two responses obtained are quite close together on the dial, and many people, including engineers, have operated a FreModyne receiver for several minutes before realizing that there was more than one response.

The f-m signal picked up by the antenna is applied through a signal-frequency tuned circuit to the grid of the superregenerator. Here it is mixed with the local-oscillator signal, produced by a conventional Colpitts oscillator circuit.



THE FREMODYNE SUPERREGENERATIVE-SUPERHETERODYNE CIRCUIT

The resulting 22-megacycle signal is amplified by a Colpitts-oscillator type of superregenerative detector, and audio is recovered across a 22000-ohm resistor in the lead from cathode to B minus. After filtering out quench and applying de-emphasis, the audio signal is delivered, ready to be fed to a conventional audio amplifier. A resistor of 1500 ohms and a capacitor of 2500 $\mu\mu f$ control the quench wave shape. Another resistor of 150,000 ohms and an electrolytic capacitor of 10 μf permit stabilized operation with a large audio output.

A low-priced a-m/f-m receiver using the FreModyne circuit can be obtained by adding the single FreModyne double triode to a conventional four-tube-plus-rectifier a-m receiver. The FreModyne circuit then merely uses the audio amplifier and power supply of the a-m set. This arrangement permits very simple switching of the audio and plate-voltage supply when changing from a.m. to f.m.

The usable f-m sensitivity of the FreModyne receiver in its present stage of development is represented by the quieting sensitivity of the order of 74 db below one volt (200 μv) and not by the maximum sensitivity (which includes values with unusable signal-to-noise ratio). A signal weaker than 74 db can be heard but at a correspondingly poorer signal-to-noise ratio. For example, an 83-decibel (70-μv signal gives approximately 20 db signal-to-noise ratio. The amount of radiation is considerably less than that of conventional superregenerative receivers and somewhat less than many conventionally designed medium-priced f-m receivers.

The selectivity of the FreModyne circuit is better than that of many conventionally designed receivers. It is sufficient for good rejection of local stations, particularly when the receiver is side-tuned on the appropriate side of the desired signal, that is, away from the interfering signal.

The FreModyne circuit discriminates against impulse noise such as due to automobile ignition. The use of superregeneration makes the receiver periodically sensitive for short intervals, so that it completely ignores many impulses occurring between these intervals. The detector characteristic is logarithmic so that the large-amplitude noise pulses that are not ignored, are crushed or compressed.

Typical F-M Performance Data for A-M/F-M Set Using FreModyne Circuit

Values are averages for several Hazeltine and licensee developmental receivers.

Test Conditions:

Frequency: average performance over 88-108 band
Dummy Antenna: 300 ohms
Standard Output: 0.05 watt
Standard Modulation: ±22.5 kc at 400 cycles

Quieting Sensitivity (*S/N=30 db*) **74 db (200 μv)**
Maximum Sensitivity (*using 400-cycle filter*) **102 db (8 μv)**
Audio Power Output (*max. vol.—60 db input*) **1 watt**

Radiation:

Oscillator Frequency 14 μwatts (65 mv r-m-s)
Signal Frequency (a short pulse of 10% duty cycle)
Peak Pulse Power 19 μwatts (75 mv r-m-s)
Average Power 1.9 μwatts

OPTIMUM FILTER TERMINATIONS

● Although classical filter theory specifies the constants of the various components as a function of the terminating resistance, it has long been understood that varying effects could be obtained by a change of this value. L. J. Giacoletto reports an interesting study of this subject in the September 1947 issue of *RCA Review*, under the title "Optimum Resistance Termination for Single-Section Constant-*k* Ladder-Type Filters."

The optimum termination resistance for a desired operating characteristic from a single section of a constant-*k* ladder-type filter permits improved performance without extra circuit elements. Sections may be designed in the usual manner with the substitution of the value for R which is equal to R_L/k, where R_L is the impedance of the generator, and that for which the filter would normally

[*Continued on page 38*]

Two-Way Speaker

[*from page 19*]

to below 40 cps, the resulting curve is then inspected to determine if the two peaks are of equal amplitude. If they are, the construction is completed, and no further adjustment will have to be made to the box. If, however, the two peaks are dissimilar, experiment with a piece of wood in front of the port until they are alike. Noting the amount of the port area that is covered by the external wood, remove the back and install an equal sized block of wood on the inside of the cabinet over the opening, with another piece of expended metal covering the port. Then finish the lining of the front panel.

It may possibly be found that the port is too small, and it will then be necessary to enlarge it to obtain the desired result. However, this is a matter of experimentation, and will vary with each speaker used. No specific instructions can possibly be given unless the exact dimensions of the cabinet are known, together with the resonant frequency of the low-frequency speaker.

When the port dimensions are completely determined, the exterior of the box may be painted, stained, or finished in any way desired. However, since the high-frequency unit and horn are to be mounted on top of it, it may be that some other housing must be provided in order for the entire speaker to have a passable eye appeal. *Fig. 3* shows a typical arrangement for the complete speaker, without any provision for improving the appearance. Before going further with the steps necessary to complete the external appearance, it is desirable that the final design be determined. The complete speaker is to consist of the reflex cabinet for the low-frequency speaker, the high-frequency unit and horn, and a dividing network. The horn must be correctly mounted in order for the results to be at all passable, and the remaining steps for the completion of the two-way speaker system will be covered in the concluding article of this series, which is scheduled for the January issue.

Wide-Range Reproduction

[*from page 19*]

examine this matter of Cidental and Incidental listening. Outside of a few organ and tympani players, even musicians will not miss the two very costly end octaves missing in the system described even while listening cidentally. However, cidental listening amounts to only a small percentage on radio music under the best of conditions. Even with Toscanini or Waring, probably only between 10 and 30 per cent of *their* audiences listen cidentally. On dialogue shows like Hope or Lux, the percentage will be higher, but on dialogue an octave could be removed from each end of the system defined without serious damage to articulation and enjoyment. On the vast majority of network radio shows, the percentage of cidental listening probably falls well below 10 per cent. Probably, over a year's time for all listeners and all shows, the average percentage of cidental listening is somewhere around 1 per cent. In the case of home reproduction of records, the percentage of cidental listening is probably higher for classical records and lower for popular records than the estimates given for radio reception. However, the sales of classical records are so very much less than for popular records that even an assumption of 100 per cent cidental listening to classical records will not change the over-all picture more than a few per cent. The fact is that *most* listeners *usually* listen *incidentally*.

System Requirements

What difference does the listener require in the system between cidental and incidental listening? For cidental listening, practically all listeners would probably prefer high fidelity, as defined above. Incidental listening requires a degraded frequency response and volume range. For incidental listening, probably all listeners prefer music to dialogue, although a few lonely people may prefer a background mumble of voices to make them feel less alone.

The important factor in discussing system degradation for incidental listening is the matter of "masking." By masking is meant not only the masking of hearing but the masking of visual perception, attention, and thought itself. It is probably that all these factors vary closely with the masking of hearing alone.

There seems good reason to believe that the masking of dialogue by music or noise is closely related to the intelligibility of the unvoiced consonants such as *s, f, c, g,* and *t*. The most important octave involved in masking these sounds is that from 2,000 to 4,000 cps. Thus it appears that the loudness of the masking sound in the 2,000 to 4,000 cps region determines the major factor in the "Irritation Index" of incidental music as a background to dialogue. The Juke Box manufacturers recognize this factor by starting down from about 1,500 cps at a rate of better than 10 db per octave. A better scheme might be to attenuate the 2,000 to 4,000 cps octave, then to go back up again to catch the weak high-frequency sounds which help so much in separating and identifying individual musical instruments.

A simple two-position tone control marked Cidental-Incidental might satisfy the majority of listeners.

Musical Acoustics

[*from page 26*]

by an analysis of its harmonics irrespective of their respective phases. That is not true with a graph. As is shown in *Fig. 13*, the same two harmonics producing the same tone of the same loudness, when combined at different phases produce graphs of different form, different maximum amplitudes, and with a different number of maximum peaks per cycle.

Furthermore, graphs showing equal amplitudes do not represent tones of equal loudness. The energy or loudness varies directly as both the square of its frequency and as the square of its amplitude. Therefore, the casual inspection of a sound track may be deceptive, and the prediction of minimum spacing required between sound tracks for a certain tone cannot be made except by harmonic analysis. This is illustrated in *Fig. 14*. It shows the imposition of the 10th harmonic with relative amplitude of 0.30 upon a fundamental of relative frequency 1, amplitude 1, and energy 1. The relative energy of the harmonic is n^2a^2 or 9.00 and although the relative amplitude of the complex resultant tone is only 1.265 and the relative squares of amplitudes is only 1.60, yet its energy (loudness) is 10.00.

This but illustrates the well-known fact that the most energy in a complex tone does not necessarily lie in its low frequencies, and that the low frequency tones control the minimum width of spacing between sound grooves. And it also points to the importance of the microscopic small amplitude irregularities in a sound groove. They add extraneous high frequencies and provoke the natural resonant frequencies of the recording and reproducing device.

With cut grooves microscopic fissures result from the tearing away of the thread of material and burnishing is resorted to for curing such defects. The uneven width of the sound groove places a vertical thrust upon the playback stylus at the "pinches" to promote foreign vibrations; and permits the stylus to rattle around at the crests of waves.

However, the problem of maximum slopes has been approached from the consideration of simple sine curves and it is far more difficult to conceive how any sound reproducing system using physical contacts and restraints can record and reproduce the still more steeply-pitched curves resulting from the integration of some possible series of harmonics. Two extreme cases are shown in *Figs. 15a* and *15b*, both of which approach and reach at their limit a series of slopes at 90° to the unmodulated sound track; and they would offer a direct abutment to an amplitude-controlled pickup.

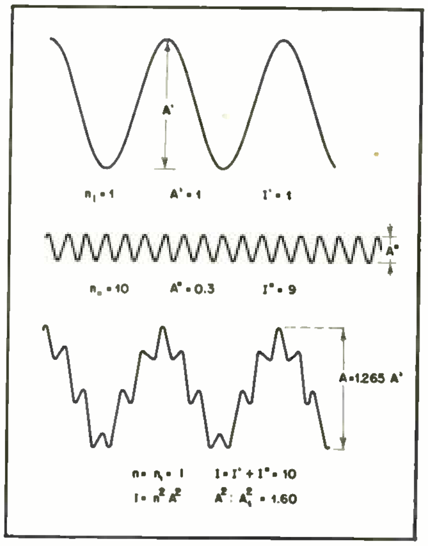


Fig. 14. Complex Wave.

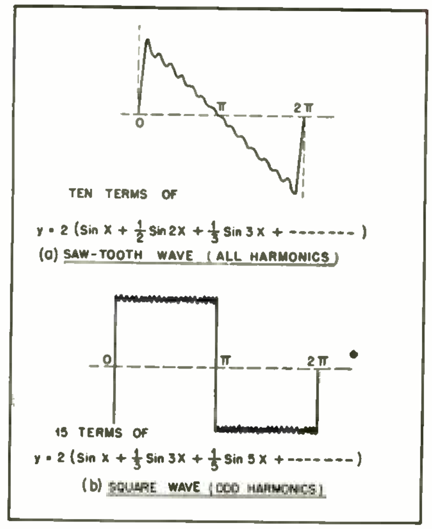


Fig. 15A. Saw-tooth Wave. B. Square Wave.

If all the even and odd harmonics are present with respective amplitudes varying inversely as their frequencies a saw-tooth form of curve results, having the equation $y = 2\,[\sin x + \frac{1}{2}\sin 2x + \frac{1}{3}\sin 3x +]$ This is shown in *Fig. 15a* where only the first ten terms of such a series have been added. With an infinite number of terms it would show vertical straight lines connected by straight diagonals.

Where only the odd harmonics are included in such a series a square wave form results, and its equation is

$$y = 2\,[\sin x + \tfrac{1}{3}\sin 3x + \tfrac{1}{5}\sin 5x + \text{——}]$$

Fig. 15b shows such a curve for the first 15 terms of that equation. With an infinite number of odd harmonic terms its form would show vertical straight lines connected by horizontal straight lines.

Record Revue

[*from page 27*]

itol calls this material, "Unbreakable under normal conditions." The record is a none too humorous children's story full of sound effects and animal noises.

Bread and Butter Woman Cy Oliver and his Orchestra . . . . MGM 10103
One more phony Calypso number with lyrics far less amusing than authentic West Indian songs. The vocal, by Oliver and a chorus, is well recorded, but the band sounds muffled. The range of the record does not exceed the limits of the poorest reproducing equipment.

[*Edward Tatnall Canby's column will reappear next month, as well as more by Bertram Stanleigh.*]

Impedance Matching

[*from page 22*]

these reflections will cause standing waves on the line which may produce voltage nodes of a value considerably higher than the voltage applied at the source end. This may result in cross-talk into other lines. The very long line also involves loading the line with inductances and a number of other complicated corrections.

Long Lines

Fortunately, in sound distribution systems, long lines are seldom used. Perhaps it might be well to define what is a "long" line. A transmission line is long when the length of the line becomes comparable to one-quarter wavelength of the highest frequency it is desired to transmit. Assuming an open wire line and the approximate assumption that the velocity of transmission equals the speed of light, 10,000 cps has a wavelength of 18.6 miles. Therefore, for transmitting 10,000 cps, a line more than 4 or 5 miles long is a long line.

For a line 1 to 5 miles long, the problem is more complicated than for the short line but far easier than for the long line. It is not necessary to terminate in the characteristic impedance, although it is necessary to terminate in an impedance of the same order of magnitude as the characteristic impedance. Furthermore, it is necessary, if the line is rented from a telephone company, to abide by the standard of + 8 vu for the terminal volume level. If this value is exceeded, the line may cross-talk into adjacent lines. If the level falls very far below this value, the signal-to-noise ratio will be impaired. It may be necessary, depending upon the length of the line, to make correction for line losses. *Figures 2* and *3* show that losses in either open wire or cable lines become greater with frequency.

For wired-music applications, a common practice is to provide a termination of about 150 ohms at the central distributing point. The line runs through a one-to-one isolating transformer into the telephone exchange where it may branch to other telephone exchanges and then may branch again to several subscriber locations. This is in effect a moderate-length line with a 150-ohm termination on one end and a number of subscriber amplifiers on the other end. If the lines are not more than about five miles long, it is perfectly feasible to place bridging amplifiers with high input impedances at the far end. Of course, the telephone company may require an isolating transformer ahead of each amplifier, but the line is terminated in an impedance that is approximately equal to the input impedance of a subscriber amplifier divided by the number of amplifiers. This should turn out to be a value of the same order of magnitude as the termination on the other end of the line. If the program level on the telephone line is restricted to + 8 vu, no great harm will result from an imperfect match. If the lines run much further than this, however, it is preferable that each line be properly terminated at both ends in a value of impedance equal to the characteristic impedance of the line.

Magnetic Recording

[*from page 16*]

point where the gap becomes so narrow that efficiency again suffers from flux leakage across the gap. Obviously zero gap opening would make a very inefficient reproducing head.

Conclusion

Over wide limits, the gap width of the recording head has little effect on the recording. It should be sufficiently wide so that strong fields have the opportunity to penetrate the tape yet not so wide that excessive currents are required for excitation. The frequency response of the playback system is definitely affected by the gap width of the reproducing gead. The demagnetizing forces developed in a recording attenuate the induction at high frequencies.

In Part III of this article, we shall combine these effects and consider their influence on the over-all response. Attention will then be given equalization, noise, distortion, signal levels and the effects of velocity of the tape drive.

Sound Reinforcement

[*from page 7*]

speakers to radiate the higher frequencies with directional properties, usually in the form of a beam, the width of the beam being a function of the loudspeaker design and the frequency involved. These directional properties permit the loudspeakers to be located in such a manner that the microphones are protected from receiving appreciable direct pickup from the loudspeaker units.

The problem of feedback may appear to be complicated by the presence of "audience reaction" microphones in the audience section of the studio. However, difficulty with feedback will not be encountered from this source provided no attempt is made to reinforce the output of these microphones. Another difficulty, commonly known as "slap-back," may be encountered when the audience reaction microphones are open. This is due to sound from the loudspeakers being picked up by the audience reaction microphones and appearing as an echo on the air channel. The only means of

reducing slap-back is to protect the audience reaction microphones from the full field of the loudspeakers by careful placement of both microphones and loudspeakers.

Conclusion

The sound reinforcement system design details discussed above have been employed in various studios and radio theaters of the Columbia Broadcasting System and have proven satisfactory from both an engineering and operational viewpoint. Many of the ideas discussed are the result of suggestions contributed by members of the CBS staff over a period of years. Although it is not possible to associate any specific idea with a given individual, the contributions of all are gratefully acknowledged.

Technicana

[*from page 33*]

be designed. The optimum value of k for various characteristics of the completed filter may be determined by making a series of calculations for different values and determining the Standard Error (S.E.) for these values. By the application of the calculus, it is possible to select the maximum or minimum value of the S.E. for a particular desired characteristic, or it may be determined by graphical means with greater ease.

This determination of the Standard Error is made over the pass band, and when such desired characteristics are optimized, it is probable that the performance may be affected adversely outside the pass band, although such changes are usually small compared to those within the pass band.

By this method, it is possible to adjust filter characteristics to provide any of the following: optimum uniformity of resistive impedance throughout the pass band; minimum reactive impedance; linear phase shift; and maximum power transfer. The article carries a table showing the values of k which provide these characteristics.

VOLTAGE REGULATORS

• Voltage regulated power supplies of the type using a controlled series tube are discussed with considerable thoroughness by Leonard Mautner in the September issue of *Electrical Engineering*, and a number of important considerations which may not be obvious to the designer of such equipment are brought out.

Primarily, this is a study of the basic series type, shown in *Fig. 1*, and it reduces the operation of such circuits to mathematical precision. This type of regulator has some points of similarity to a cathode follower, and in itself tends to attenuate variations in the plate voltage. To some extent this alone would be an improvement without the regulating control circuit.

The ratio of the change in the voltage in the output circuit to that at the plate of V_1 is equal to $1/\mu G\beta$ when the load resistor for the plate of the control tube is fed from the cathode of the series tube instead of from the plate. In ordinary single-stage control circuits, the value of G approximates 100, and the μ applies to the series tube, V_1. An expansion of the control amplifier will provide a considerable improvement in the ratio so that with the "cascade" amplifier it is of the order of 0.2 per cent. Using a difference amplifier in the control circuit will provide four times better regulation, with a ratio of about 0.05 per cent. With a two-stage amplifier, the ratio may increase to 0.013 per cent, and with a three-stage amplifier the ratio may be increased to about 0.007 per cent.

When such high degrees of regulation are required, it may become necessary to operate the amplifier heaters from a d-c source, or in elaborate designs in which a large amount of power is furnished, it may be possible to employ the

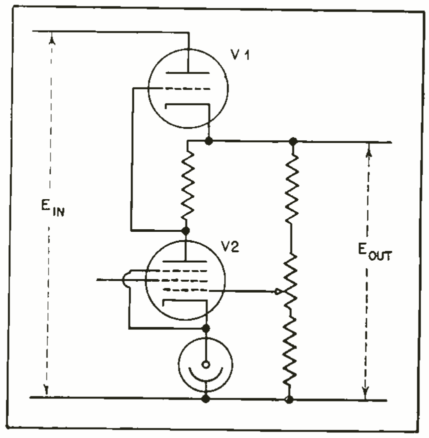


Fig. 1

output voltage in a series filament string for the regulator amplifier, thus providing the advantage of both d-c supply to the filaments giving further reduction to the possibility of hum, and a regulated voltage supply maintaining a more constant d-c output voltage free from fluctuations caused by variations in the emission of the amplifier tubes.

CREDIT

• We regret exceedingly our failure to credit the co-author of the paper detailing the advantages of the Space-Charge-Grid Output Tubes, a report of which was presented in the October issue. While Norman C. Pickering presented the paper after constructing and measuring the amplifier circuits employing the tubes, the actual development of the tubes themselves was carried out by W. S. Brian, and he collaborated with Mr. Pickering in preparing the paper. In addition, he conducted the discussion of the paper following its presentation.

Letters

[from page 3]

building. A search revealed the fact that several piles of stacked piping laid out horizontally were causing the peculiar acoustic absorption effects. It occurred to us that perhaps such cylindrical "arrays" or "honeycombs" might be of considerable use in controlling studio acoustics.

As for *Audio Engineering*—all we can say is—it's about time! In looking back on "revolutions" in the various science and engineering fields, one usually finds some key man or circumstance or factor that was mainly responsible. In your journal, we see the beginnings of such an upheaval in a field which has long been crippled by a maze of contradictions, antiquated notions, fallacies and misinformation unparalleled in all the fields of communication and sound transmission.

Congratulations!

Ted Powell
5719 - 69th Avenue
Maspeth, New York

New Products

[from page 31]

Output. 60 millivolts at 1000 cps with lateral displacement of .001″.

Impedance. Standard cartridge, high impedance; other impedances available on request.

Electric Characteristics. Inductance 350 millihenries at 1000 cps; "Q" 1.05; d-c resistance, 1450 ohms.

Mounting. Standard mounting holes, ½″ between centers 3-48 screws.

Weight. 30 grams.

This pickup is manufactured by the Clarkstan Corporation, 11927 West Pico Boulevard, Los Angeles 34, Calif.

New I. R. E. President and Vice-President

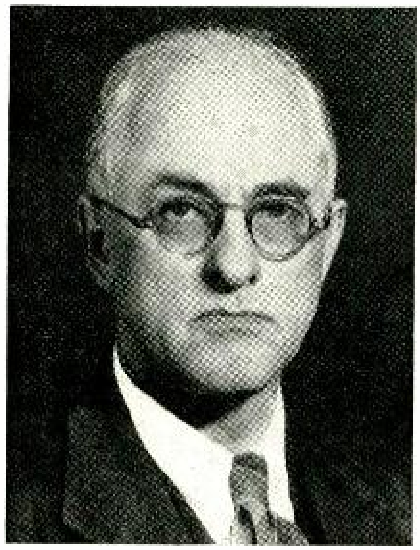
• The Institute of Radio Engineers has announced the election of Benjamin E. Shackelford as president of the Institute for the year 1948. Dr. Reginald L. Smith-Rose of England was elected Vice President.

Dr. Shackelford has been a Fellow of the I.R.E. since 1938. He is manager of the license department of RCA international division, New York, N. Y. Dr. Smith-Rose, a Fellow of the I.R.E. since 1944, is superintendent of the radio division, National Physical Laboratory, Teddington, Middlesex, England.

This Month

MAXFIELD BECOMES CONSULTANT

● Joseph P. Maxfield, internationally known pioneer in research and the practical development of sound transmission, recording and reproduction, whose retirement from the Bell Telephone Laboratories was re-



Joseph P. Maxfield

cently announced, has become associated with the Altec Lansing Corporation, as a Consulting Engineer, according to a statement issued by G. L. Carrington, President of Altec Lansing. Mr. Maxfield is a member of the Advisory Board of Audio Engineering.

FTR APPOINTS BLAYLOCK

● It has been announced by Federal Telephone and Radio Corporation, Clifton, N. J., that Captain L. B. Blaylock, U. S.

Capt. L. B. Blaylock, U.S.N. (Ret.)

Navy (retired), who recently joined the company, has been appointed Director, Radio Division. FTR is the American manufacturing affiliate of International Telephone and Telegraph Corporation.

PICKERING APPOINTS MORHAN

● Morhan Exporting Corp., 458 Broadway, New York City, has just been appointed exclusion export representatives for Pickering Co., Inc., manufacturers of the Pickering pick-up, equalizers, and cartridge reproducers.

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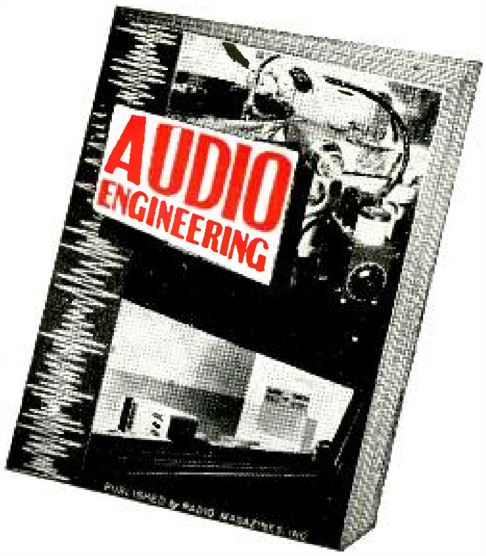

Each issue covers 5 or more of these subjects

1—Broadcasting

2—Transmitter and receiver manufacturing

3—Sound-on-film equipment

4—Recording (disc, wire and tape)

5—Public address

6—Industrial sound equipment and applications

7—Acoustic treatment of studios, rooms, auditoriums, etc.

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HOWARD A. CHINN
Chief audio engineer for the Columbia Broadcasting System

J. P. MAXFIELD
Authority on sound engineering, Bell Telephone Labs.

JOHN D. COLVIN
Audio Facilities Engineer, American Broadcasting Company

GEORGE M. NIXON
Ass't Director of Technical Development
National Broadcasting Company

who are members of the editorial advisory board.

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